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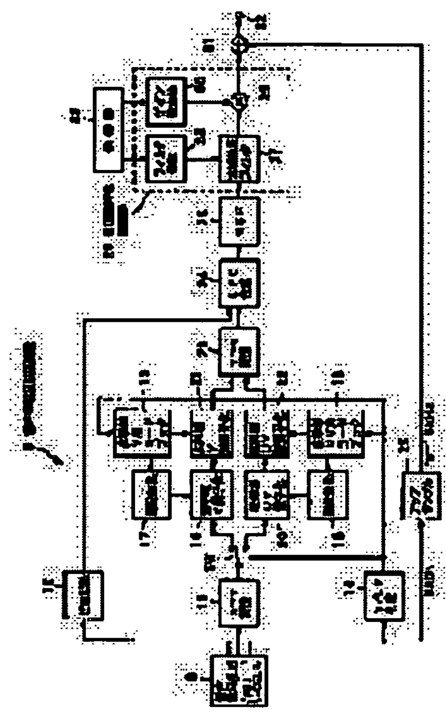
(54) METHOD AND SYSTEM FOR EXTENDING BANDWIDTH

(57)Abstract:

PROBLEM TO BE SOLVED: To provide a method and system for extending bandwidth with which a frequency characteristic of a high frequency component of a broadband signal is adjusted, in matching with the preference of the user.

SOLUTION: A frequency characteristic adjustment section 26 of a voice band width extension device 9 adjusts the frequency characteristic of a high frequency component of 3,400 Hz or over from a BSF 25, based on a parameter that is given in advance and can be revised. An adder 31 adds a frequency component of 3,400 Hz or over with an adjusted frequency characteristic to an original narrow band voice component with a frequency band of 300 Hz-3400 Hz from an up-sample circuit 25.



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CLAIMS

[Claim(s)]
[Claim 1] The bandwidth escape approach characterized by adding to the above-mentioned narrow-band signal after the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band, and which can be changed adjusts in the bandwidth escape approach which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this.
[Claim 2] The bandwidth escape approach according to claim 1 characterized by adjusting the gain of the above-mentioned component out of band as adjustment of the above-mentioned frequency characteristics.
[Claim 3] The bandwidth escape approach according to claim 1 characterized by adjusting the frequency band of the above-mentioned component out of band as adjustment of the above-mentioned frequency characteristics.
[Claim 4] The bandwidth escape approach characterized by adjusting the frequency characteristics of the above-mentioned component out of band after being added to the above-mentioned narrow-band signal with the parameter value which was able to be given beforehand, and which can be changed in the bandwidth escape approach which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this.
[Claim 5] The bandwidth escape approach according to claim 4 characterized by adjusting the frequency band of the above-mentioned component out of band as adjustment of the above-mentioned frequency characteristics.
[Claim 6] The bandwidth growth equipment characterized by to have the frequency-characteristics adjustment device which adjusts with the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band, and which can be changed in the bandwidth growth equipment which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, and an addition means add the component out of band to which frequency characteristics were adjusted with the above-mentioned frequency-characteristics adjustment device to the above-mentioned narrow-band signal.
[Claim 7] The above-mentioned frequency-characteristics adjustment device is bandwidth growth equipment according to claim 6 characterized by adjusting the gain of the above-mentioned component out of band.
[Claim 8] The above-mentioned frequency-characteristics adjustment device is bandwidth growth equipment according to claim 7 characterized by carrying out the multiplication of the GEINN set point which was given beforehand, and which can be changed to the above-mentioned component out of band.
[Claim 9] The above-mentioned frequency-characteristics adjustment device is bandwidth growth equipment according to claim 6 characterized by adjusting the frequency band of the above-mentioned component out of band.
[Claim 10] The above-mentioned frequency-characteristics adjustment device is bandwidth growth equipment according to claim 9 characterized by being given beforehand and adjusting the frequency band of the above-mentioned component out of band based on the filter factor in which ****** is possible.
[Claim 11] The bandwidth growth equipment which carries out [having the frequency-characteristics adjustment device which adjusts with the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band of the addition outputs of an addition means add the above-mentioned component out of band to the above-mentioned narrow-band signal, and the above-mentioned addition means, in the bandwidth growth equipment which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, and which can be changed, and] as the description.
[Claim 12] The above-mentioned frequency-characteristics adjustment device is bandwidth growth equipment according to claim 11 characterized by adjusting the frequency band of the above-mentioned component out of band of the addition outputs of the above-mentioned addition means.
[Claim 13] The above-mentioned frequency-characteristics adjustment device is bandwidth growth equipment according to claim 12 characterized by adjusting the frequency band of the above-mentioned component out of band of the addition outputs of the above-mentioned addition means based on the filter factor which was able to be given beforehand, and which can be changed.

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DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Field of the Invention] This invention leaves the parameter which constitutes the narrow sound signal or narrow it of a frequency band which is told by a communication link and broadcast as it is in transmission and a transmission line, and relates to the bandwidth escape approach and equipment which extend bandwidth by the receiving side and are made into a wideband voice signal. Moreover, it is related with the bandwidth escape approach and equipment which extend the bandwidth of the signal accumulated in package media, and are made into a broadband signal.
[0002]
[Description of the Prior Art] The band of the telephone line is as narrow as 300-3400Hz, and the frequency band of the sound signal sent through the telephone line is restricted. For this reason, the tone quality of the conventional analog telephone line can seldom be said to be fitness. Moreover, there is dissatisfaction also about the tone quality of a digital cellular phone.
[0003] However, since the specification of a transmission line has become settled, it is difficult to extend this bandwidth. Therefore, a signal component out of band is predicted by the receiving side, and the various proposal of the system which generates a broadband signal is made.
[0004] Vector sum exciting line form prediction (Vector Sum Excited Linear Prediction:VSELP) coding which is the voice codec method of the automobile/cellular phone of our country especially, By the method which tried application to the source of pitch synchronous noise excitation-sign exciting line form prediction (Pitch Synchronus Innovation-CodeExited Linear Prediction:PSI-CELP) coding method, it notes performing LPC composition. Both linear predictor coefficients alpha and the source of excitation are broadband-ized, and there are broadband-ized alpha and a thing which performs LPC composition by the source of excitation.
[0005] However, distortion is included in the wideband voice obtained by this. Then, in the frequency component contained in the Hara voice, since it is [with the Hara voice] naturally more nearly quality, the filter removed this component among the compounded wideband voices, and how to add the Hara voice has been taken.
[0006]
[Problem(s) to be Solved by the Invention] By the way, although it was the wideband voice compounded as mentioned above, the favorite individual difference of tone quality was large, and the gain of the high-frequency component by which guess composition was carried out was understood that how to bend to a fixed value is good. Similarly, a high-frequency component 6kHz or more has desirable how depending on which this value also bends to immobilization, although the way oppressed a little is liked.
[0007] This invention is made in view of the above-mentioned actual condition, and aims at offer of the bandwidth escape approach and equipment which can adjust the frequency characteristics of a high-frequency component according to liking of a user.
[0008]
[Means for Solving the Problem] Since how to add the Hara voice and the compounded component out of band about gain is taken, it becomes possible by adjusting the gain of a component out of band before addition. Moreover, it becomes possible by covering the filter which adjusts frequency characteristics before addition or after addition about bandwidth.
[0009] For this reason, in the bandwidth escape approach which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, after the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band and which can be changed adjusts the bandwidth escape approach of this invention, it is added to the above-mentioned narrow-band signal.
[0010] Moreover, in the bandwidth escape approach which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, the parameter value which was able to be given beforehand and which can be changed adjusts the frequency characteristics of the above-mentioned component out of band after being added to the above-mentioned narrow-band signal.
[0011] Furthermore, the bandwidth growth equipment of this invention is equipped with the frequency-characteristics adjustment device adjusted with the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band, and which can be changed, and an addition means add the component out of band to which frequency characteristics were adjusted with the above-mentioned frequency-characteristics adjustment device to

the above-mentioned narrow-band signal, in the bandwidth growth equipment which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this.
[0012] moreover, the above of the addition outputs of an addition means add the above-mentioned component out of band to the above-mentioned narrow-band signal in the bandwidth growth equipment which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, and the above-mentioned addition means -- even if few, it has the frequency-characteristics adjustment device which adjusts with the parameter value which was able to give beforehand the frequency characteristics of one component out of band, and which can be changed.
[0013]
[Embodiment of the Invention] Hereafter, it explains, referring to a drawing about the gestalt of operation of this invention. The gestalt of this operation is speech bandwidth growth equipment which extends the bandwidth of the inputted narrow-band voice, using the bandwidth escape approach concerning this invention. The bandwidth escape approach which this speech bandwidth growth equipment uses is the bandwidth escape approach which guesses a component out of band, adds to the narrow-band signal compounded from the parameter, and extends bandwidth from the parameter which can compound the narrow-band signal restricted in a transmission line, and after the parameter value to which the frequency characteristics of the above-mentioned component out of band were beforehand given by the request of a user and which can be changed adjusts it, it is the approach of adding to the above-mentioned narrow-band signal. For details, it mentions later.
[0014] This speech bandwidth growth equipment is applied to a digital cell phone unit. First, the configuration of this digital cell phone unit is explained. Here, although the transmitter and receiver side is described separately, it is collectively built in one cell phone unit in fact.
[0015] In a transmitter side, the sound signal inputted from the microphone 1 is changed into a digital signal with A/D converter 2, after encoding with the voice encoder 3, transmitting processing is performed to an output bit with a transmitter 4, and it transmits from an antenna 5.
[0016] At this time, the voice encoder 3 supplies the coding parameter in consideration of narrow-band-ization restricted by the transmission line to a transmitter 4. For example, as a coding parameter, there are a parameter about the source of excitation and linear predictor coefficients alpha.
[0017] Moreover, in a receiver side, a receiver 7 receives the electric wave caught with the antenna 6. And the above-mentioned coding parameter is decoded with the voice decryption vessel 8, and voice is extended using the above-mentioned decryption parameter with speech bandwidth growth equipment 9. Then, it returns to an analog sound signal with D/A converter 10, and outputs from a loudspeaker 11.
[0018] The 1st example of the above-mentioned speech bandwidth growth equipment 9 in this digital cell phone unit is shown in drawing 2 . The speech bandwidth growth equipment 9 shown in this drawing 2 extends audio bandwidth using the coding parameter sent from the voice encoder 3 of the transmitting side of the above-mentioned digital cell phone unit.
[0019] The above-mentioned coding parameter is decoded with the voice decryption vessel 8. If the coding approach in a voice coder 3 is based on a PSI-CELP (Pitch Synchronus Innovation-CELP: source of pitch synchronous noise excitation-CELP) coding method, the decryption approach in this voice decryption machine 8 is also depended on PSI-CELP.
[0020] The parameter about the source of excitation which is the 1st coding parameter of the above-mentioned coding parameters decoded with the voice decryption vessel 8 is supplied to the zero stuffing section 12. Moreover, the linear predictor coefficients alpha which are the 2nd coding parameter of the above-mentioned coding parameters are supplied to the alpha->r (linear predictor coefficients -> autocorrelation) conversion circuit 13. Moreover, the signal decoded with the voice decryption vessel 8 is supplied to the V/UV judging circuit 14.
[0021] Moreover, speech bandwidth growth equipment 9 is equipped with the code book 15 for broadband voiced sounds and the code book 16 for broadband silent sounds which are beforehand created using the object for voiced sounds and the parameter for non-vocal sound which were extracted from a broadband voiced sound besides the zero stuffing section 12, the alpha->r conversion circuit 13, and the V/UV judging circuit 14, and non-vocal sound.
[0022] Furthermore, the partial extract circuit 17 and the partial extract circuit 18 which this speech bandwidth growth equipment 9 carries out the partial extract of each code vector in the code book 15 for broadband voiced sounds, and the code book 16 for broadband silent sounds, and ask for a narrow-band parameter, The quantizer 19 for narrow-band voiced sounds which quantizes the autocorrelation for narrow-band voiced sounds from the alpha->r conversion circuit 13 using the narrow-band parameter from the partial extract circuit 17, The quantizer 20 for narrow-band silent sounds which quantizes the autocorrelation for narrow-band silent sounds from the above-mentioned alpha->r conversion circuit 13 using the narrow-band parameter from the partial extract circuit 18, The reverse quantizer 21 for broadband owner vocal sound which reverse-quantizes the narrow-band voiced sound dosage child-ized data from the quantizer 19 for narrow-band voiced sounds using the code book 15 for broadband voiced sounds, The reverse quantizer 22 for broadband non-vocal sound which reverse-quantizes the narrow-band silent sound dosage child-ized data from the quantizer 20 for narrow-band silent sounds using the code book 16 for broadband silent sounds, While changing the autocorrelation for broadband voiced sounds used as the reverse quantization data from the reverse quantizer 21 for broadband owner vocal sound into the linear predictor coefficients for broadband voiced sounds The autocorrelation which changes the autocorrelation for broadband silent sounds used as the reverse quantization data from the reverse quantizer 22 for broadband non-vocal sound into the linear predictor coefficients for broadband silent sounds -> The linear-predictor-coefficients (r->alpha) conversion circuit 23, It comes to have the LPC composition circuit 24 which compounds wideband voice based on the linear predictor coefficients for broadband voiced

sounds from this r->alpha conversion circuit 23, the linear predictor coefficients for broadband silent sounds, and the source of excitation from the zero stuffing section 12.
[0023] Moreover, this speech bandwidth growth equipment 9 is equipped with the band stop filter (BSF) 25 which removes the signal component of 300Hz - 3400Hz of frequency bands of input narrow-band voice data from the synthetic output from the rise sample circuit 25 which carries out over sampling technique of the sampling frequency of the narrow-band voice data decrypted with the voice decryption vessel 8 to 16kHz from 8kHz, and the LPC composition circuit 24.
[0024] Furthermore, this speech bandwidth growth equipment 9 is equipped with the frequency-characteristics controller 26 which adjusts the frequency characteristics of the high frequency component 3400Hz or more from BSF25 with the parameter value which was able to be given beforehand, and which can be changed, and the adder 31 which adds the frequency component 3400Hz or more to which frequency characteristics were adjusted by this frequency-characteristics controller 26 to the narrow-band voice data component of the origin of 300Hz - 3400Hz of frequency bands from the above-mentioned rise sample circuit 25.
[0025] And from an output terminal 32, a frequency band is 300-7000Hz, and the digital sound signal whose sampling frequency is 16kHz is outputted.
[0026] Here, the frequency-characteristics controller 26 adjusts the frequency band of the above-mentioned component out of band with the high region oppression filter 27. The high region oppression filter 27 shall be a filter which oppresses component about 6kHz or more, and shall tend to hear the above-mentioned component out of band. The filter factor maintenance memory 28 is connected to the high region oppression filter 27. Some filter factors which make attenuation of frequency characteristics gently-sloping, or are made steep are memorized by this filter factor maintenance memory 28. These filter factors are chosen according to actuation by the user on a control unit 33. And with the high region oppression filter 27, the frequency band of a component out of band is adjusted using the filter factor chosen according to liking of a user.
[0027] Moreover, the frequency-characteristics controller 26 adjusts the gain of the above-mentioned component out of band. The gain set point of the shoes set up beforehand is specifically memorized in the gain set point memory 30, it chooses according to the request of a user in a control unit 33, and a multiplier 29 is supplied. For this reason, in a multiplier 29, the gain of the above-mentioned component out of band can be adjusted according to a request of a user.
[0028] On the whole, this speech bandwidth growth equipment 9 operates as follows. First, a broadband parameter is presumed from a narrow-band parameter, and the wideband voice signal is searched for in the LPC composition circuit 24. And the low-pass side which is a frequency band of the Hara voice is permuted by the Hara voice after that. That is, using BSF25 as a high pass filter, it leaves only a high region, and also in this high-frequency component, a high frequency component is oppressed with the high region oppression filter 27, gain is further adjusted in the signal-processing section 29, and it is adding to the Hara voice.
[0029] Two, broadband-izing of alpha and broadband-izing of the source of excitation, are required for presumption of a broadband parameter. Moreover, it is necessary to create beforehand the code book by the autocorrelation r which is a parameter convertible into alpha and mutual for broadband-ization of alpha. Autocorrelation r is broadband-ized by quantization by this code book, and reverse quantization.
[0030] First, broadband-ization of alpha is explained. alpha is once changed into the autocorrelation r which is a parameter showing another spectral envelope which is easy to presume a high region side paying attention to being a filter factor showing a spectral envelope, broadband-izes this, and it carries out inverse transformation to alphaw from the broadband autocorrelation rw after that. Vector quantization is used for an escape. What is necessary is to vector-quantize the narrow-band autocorrelation rn and just to calculate rw which corresponds from the index.
[0031] Since fixed relation is realized so that it may mention later, that what is necessary is to prepare only the code book by the broadband autocorrelation, in a narrow-band autocorrelation and a broadband autocorrelation, a narrow-band autocorrelation can be vector-quantized by this, and a broadband autocorrelation can be found by reverse quantization in them.
[0032] About a narrow-band signal, there is relation which shows a broadband signal in the following (1) types at the band-limited thing, then a broadband autocorrelation and a narrow-band autocorrelation.
[0033]
[Equation 1]

$$\phi(x_n)=\phi(x_w\otimes h)=\phi(x_w)\otimes\phi(h) \qquad \cdots (1)$$

[0034] Here, for phi, an autocorrelation and xn are [a broadband signal and h of a narrow-band signal and xw] the impulse responses of a band limit filter.
[0035] Furthermore, the following (2) types are obtained from the relation between an autocorrelation and a power spectrum.

[0036]
[Equation 2]

$$\phi(h)=F^{-1}\left(|H|^2\right) \qquad \cdots (2)$$

[0037] Considering another band limit filter with frequency characteristics equal to the power characteristics of this band limit filter, H', then the above-mentioned (2) formula become like the following (3) types about this.
[0038]

[Equation 3]

$$\phi(h) = F^{-1}\left(|H|^2\right) = F^{-1}(H') = h' \quad \cdots (3)$$

[0039] The pass band of this new filter and the inhibition zone are equivalent to the original band limit filter, and a damping property serves as a square. Therefore, this new filter can also be called band limit filter. Consideration of this simplifies a narrow-band autocorrelation with what band-limited, the convolution, i.e., the broadband autocorrelation, of a broadband autocorrelation and the impulse response of the filter of a band limit. That is, it becomes the following (4) types.
[0040]
[Equation 4]

$$\phi(x_n) = \phi(x_w) \otimes h' \quad \cdots (4)$$

[0041] As mentioned above, if only a broadband code book is prepared in vector-quantizing a narrow-band autocorrelation, a narrow-band vector required at the time of quantization can be created by the operation, and does not need to prepare a code book beforehand from a narrow-band autocorrelation.
[0042] Furthermore, since each rw code vector has monotone reduction or the curve fluctuated gently-sloping, even if it makes it low-pass by H', there is no big change, and a direct rw code book can perform rn quantization. However, since a sampling frequency is 1/2, it is necessary to compare every other order.
[0043] By dividing into a voiced sound (V) and non-vocal sound (UV), since the still more accurate escape is possible, this is also performing the escape of alpha. In connection with this, the code book also uses two for the object for V, and UV.
[0044] Next, the escape of the source of excitation is explained. In PSI-CELP, the rise sample of the source of excitation in a narrow-band is carried out by inserting a zero value in the zero stuffing section 12, and what generated aliasing distortion is used. Although this approach is very simple, since the power of the original voice and the difference of harmonic structure are saved, it can be said that it is quality sufficient as a source of excitation.
[0045] And Broadband alpha and the source of broadband excitation which were obtained above perform LPC composition in the LPC composition circuit 24.
[0046] Moreover, since the voice by which broadband LPC composition was carried out is of inferior quality as [this], a low-pass side is permuted with the original voice SNDN of a codec output. For this reason, 3.4kHz or more of composite tone is extracted, the rise sample of the codec output is carried out to fs=16kHz by one side, and these are added.
[0047] At this time, adjustment of the gain which carries out multiplication to a high region side is enabled according to liking of a user with the multiplier 29 of the frequency-characteristics controller 26. Since the individual difference for every user is large, this value is made adjustable. That is, the value of high region side gain is beforehand set up by the input from a user, and multiplication is performed with reference to this value.
[0048] Moreover, there is the sound which is easy to hear by giving filtering which oppresses component about 6kHz or more a little with the high region oppression filter 27 of the frequency-characteristics controller 26 to a high region side before addition. This filter factor is selectable according to liking of a user. By processing with the high region oppression filter 27 using the selected filter factor, the frequency band by the side of a high region was made selectable according to liking.
[0049] However, since the processing using this high region oppression filter 26 does not affect the power characteristics by the side of low-pass, it may be performed to the component out of band under addition output of an adder 31. That is, the high region oppression filter 27 of the frequency-characteristics controller 26 may be formed in the latter part of an adder 31. Or it is also possible to give, after adding the filter which dares have influence also on a low-pass side. Wideband voice is obtained by the above.
[0050] Next, detailed actuation of this speech bandwidth growth equipment 9 is explained using the flow chart of drawing 3 .
[0051] The alpha->r conversion circuit 13 changes into Autocorrelation r the linear predictor coefficients alpha decoded with the voice decryption vessel 8 at step S1. Moreover, the signal decoded with the voice decryption vessel 8 is decoded by the V/UV judging circuit 14 at step S2, and distinction of V/UV is performed.
[0052] If a voiced sound / non-vocal sound judging flag is judged at this step S2 to be V, the switch SW which changes the output from the alpha->r conversion circuit 13 will be connected to the narrow-band voiced phonon-ized circuit 19. Moreover, if judged with UV, Switch SW will connect the output from the alpha->r conversion circuit 13 to the narrow-band silent phonon-ized circuit 20.
[0053] When UV judging circuit 14 judges the above-mentioned voiced sound / non-vocal sound judging flag to be V, in step S4, the autocorrelation r for voiced sounds from Switch SW is supplied to the narrow-band V quantization circuit 19, and it quantizes. This quantization uses the parameter for narrow-band V for which it asked at step S3 by the partial extract circuit 17 as mentioned above.
[0054] On the other hand, at step S3, when UV judging circuit 14 judges the above-mentioned voiced sound / non-vocal sound judging flag to be UV, although the autocorrelation r for non-vocal sound from Switch SW is supplied to the narrow-band UV quantization circuit 20 and it quantizes, it quantizes also here in the partial extract circuit 18 using the parameter for narrow-band UV for which it asked by the operation.
[0055] And it reverse-quantizes using the broadband V code book 15 or the broadband UV code book 16 by the broadband V reverse quantization circuit 21 or the broadband UV reverse quantization circuit 22 which corresponds, respectively at step S5, and, thereby, a broadband autocorrelation is obtained.

[0056] And a broadband autocorrelation is changed into alpha by the r->alpha conversion circuit 23 at step S6.
[0057] On the other hand, the rise sample of the parameter about the source of excitation from the voice decryption machine 8 is carried out by zero being packed by the zero stuffing section 12 between samples at step S7, and it is broadband-ized by aliasing. And this is supplied to the LPC composition circuit 24 as a source of broadband excitation.
[0058] And at step S8, the LPC composition circuit 24 carries out LPC composition of Broadband alpha and the source of broadband excitation, and the sound signal of a broadband is acquired.
[0059] However, since it does not pass to the broadband signal searched for by prediction the way things stand but the error by prediction is included, it is of inferior quality. It is better to use the original voice SNDN of a codec output (input voice) as it is about the frequency range of input narrow-band voice especially.
[0060] Therefore, filtering by step S9 of 300-3400Hz of frequency ranges of input narrow-band voice using BSF25 removes among the composite tone from the LPC composition circuit 24.
[0061] And it adds with an adder 29 at step S13 with what carried out the rise sample of the above-mentioned original voice SNDN by the rise sample circuit 25 at step S10. At this time, there is the sound which is easy to hear by filtering with the high region oppression filter 27 which oppresses component about 6kHz or more a little to a high region side at step S11. This filter factor is made selectable as mentioned above.
[0062] Furthermore, at step S12, adjustment of high region side gain is enabled according to liking of a user using the multiplier 29.
[0063] In addition, the creation of a code book used with speech bandwidth growth equipment 9 is explained here.
[0064] Creation of a code book is an approach by GLA (Generalized Lloyd Algorithm) generally known well. Fixed time amount for every 20msec(s), for example, a frame, is asked for wideband voice, and it asks for the autocorrelation to 1 Sadaji, for example, the 6th order, for every break and its frame. The autocorrelation for every frame of this is made into training data, and a 6-dimensional code book is created. At this time, distinction of a voiced sound and non-vocal sound may be performed, the autocorrelation of a voiced sound and the autocorrelations of non-vocal sound may be collected separately, and each code book may be created. In this case, although a code book is referred to during band-spreading processing at the time of the escape of alpha, also at this time, distinction of a voiced sound and non-vocal sound is performed, and a corresponding code book is used.
[0065] With speech bandwidth growth equipment 9, although the code book 12 for broadband voiced sounds and the code book 14 for broadband silent sounds are used, the creation is explained to a detail, referring to drawing 4 and drawing 5 .
[0066] First, a wideband voice signal is prepared for study and framing is carried out to one-frame 20msec(s) at step S31. Next, the classification of a voiced sound (V) and non-vocal sound (UV) is performed by investigating frame energy, the value of a zero cross, etc. in each frame at step S32.
[0067] And in a broadband voiced sound frame, the autocorrelation parameter r to the 6th order is calculated at step S33. Moreover, at step S34, it asks for the autocorrelation parameter r to the 6th order in a broadband silent sound frame.
[0068] From the 6th autocorrelation parameter of each of this frame, a broadband parameter is extracted at step S41 of drawing 5 , and the broadband V (UV) code book of a dimension 6 is created at step S42 by GLA.
[0069] As mentioned above, with the speech bandwidth growth equipment using the decryption approach by PSI-CELP, the wideband voice which a user likes mutually can be offered by making adjustable high region gain and a high region oppression filter.
[0070] Next, it explains, referring to drawing 6 about the 2nd example of the above-mentioned speech bandwidth growth equipment. Since it is equipment with which this 2nd example also extends speech bandwidth using the coding parameter sent from the voice encoder 3 of the transmitting side of the above-mentioned digital cell phone unit, the decryption according to the coding approach in the voice encoder 3 is performed.
[0071] If the coding approach in a voice coder 3 is based on a VSELP (Vector Sum Excited Linear Prediction: vector sum exciting line form prediction) coding method, the decryption approach in the voice decryption machine 8 of the preceding paragraph of this speech bandwidth growth equipment is also depended on VSELP.
[0072] The parameter about the source of excitation which is the 1st coding parameter of the above-mentioned coding parameters decoded with the voice decryption vessel 8 is supplied to the source switch section 36 of excitation of drawing 6 . Moreover, the linear predictor coefficients alpha which are the 2nd coding parameter of the above-mentioned coding parameters are supplied to the alpha->r (linear predictor coefficients -> autocorrelation) conversion circuit 13. Moreover, the decoded signal is supplied to the V/UV judging circuit 14.
[0073] Differing from the speech bandwidth growth equipment using PSI-CELP shown in above-mentioned drawing 2 is the point of having established the source switch circuit 36 of excitation in the preceding paragraph of the zero stuffing section 12.
[0074] Although PSI-CELP is performing the codec itself and processing which can be smoothly heard on audibility especially in V, when there is this [no] in VSELP, for this reason a bandwidth escape is carried out, as the noise mixed a little, it can be heard. Then, in case the source of broadband excitation is created, processing like drawing 7 is performed by the source switch circuit 36 of excitation. Processing here is only changed with the processing which showed step S87 - step S89 to above-mentioned drawing 3 .
[0075] The source of excitation of VSELP is beta by the parameter beta (long-term prediction coefficient), bL [i] (long-term filter condition), and gamma (gain) used for a codec, and c1 [i] (excitation code vector). * bL [i] + gamma Although created as * c1[i] Among these, since the former expresses a pitch component and the latter expresses a noise component, it is beta about this. * bL [i] and gamma It divides into * c1[i]. At step S87 In the fixed time amount range, since a pitch was

considered to be a strong voiced sound when the former energy is large, it progressed to YES at step S88, and the source of excitation was made into the pulse train, and in the part without a pitch component, it progressed to NO and oppressed to 0. moreover, the step S87 -- case energy is not large -- as usual -- ** -- it considered as the source of broadband excitation by carrying out, and the zero stuffing section's 12 stuffing 0 like PSI-CELP, and carrying out a rise sample to the source of narrow-band excitation created in this way at step S89. Thereby, the quality on the audibility of the voiced sound in VSELP improved.
[0076] If this processing is written by software, it will become like the following (5) types.
[0077]
[Equation 5]

$$
\begin{array}{l}
if\left(\sum_i (\beta \bullet bL[i])^2 > \sum_i (\gamma \bullet c1[i])^2\right)\{ \\
\quad if\left(\beta \bullet bL[i] > \ \left|Max(\beta \bullet bL[i])\right|\right)\{ \\
\qquad exc_{wide}[2i] = \beta \bullet bL[i]; \\
\quad \}else\{ \\
\qquad exc_{wide}[2i] = 0; \\
\quad \} \\
\}else\{ \\
\quad exc_{wide}[2i] = \beta \bullet bL[i] + \gamma \bullet c1[i]; \\
\}
\end{array}
\quad \cdots (5)
$$

[0078] And it adds with an adder 31 at step S13 with what carried out the rise sample of the above-mentioned original voice SNDN by the rise sample circuit 25 at step S92. At this time, there is the sound which is easy to hear by filtering with the high region oppression filter 27 which oppresses component about 6kHz or more a little to a high region side at step S94. This filter factor supposes that it is selectable, as mentioned above.
[0079] Furthermore, at step S95, adjustment of high region side gain is enabled according to liking of a user using the multiplier 29.
[0080] In addition, this invention is not limited only to what predicts a high region from low-pass. In a means to predict a broadband spectrum, a signal is not restricted to voice.
[0081] Moreover, also when reproducing the signal accumulated in package media with a regenerative apparatus and extending bandwidth, it can apply.
[0082]
[Effect of the Invention] According to this invention, the wideband voice suitable for liking of a user can be offered by making adjustable the frequency characteristics of a high-frequency component, for example, gain, and a frequency band.

[Translation done.]

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TECHNICAL FIELD

[Field of the Invention] This invention leaves the parameter which constitutes the narrow sound signal or narrow it of a frequency band which is told by a communication link and broadcast as it is in transmission and a transmission line, and relates to the bandwidth escape approach and equipment which extend bandwidth by the receiving side and are made into a wideband voice signal. Moreover, it is related with the bandwidth escape approach and equipment which extend the bandwidth of the signal accumulated in package media, and are made into a broadband signal.

[Translation done.]

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PRIOR ART

[Description of the Prior Art] The band of the telephone line is as narrow as 300-3400Hz, and the frequency band of the sound signal sent through the telephone line is restricted. For this reason, the tone quality of the conventional analog telephone line can seldom be said to be fitness. Moreover, there is dissatisfaction also about the tone quality of a digital cellular phone.
[0003] However, since the specification of a transmission line has become settled, it is difficult to extend this bandwidth. Therefore, a signal component out of band is predicted by the receiving side, and the various proposal of the system which generates a broadband signal is made.
[0004] Vector sum exciting line form prediction (Vector Sum Excited Linear Prediction:VSELP) coding which is the voice codec method of the automobile/cellular phone of our country especially, By the method which tried application to the source of pitch synchronous noise excitation-sign exciting line form prediction (Pitch Synchronus Innovation-CodeExited Linear Prediction:PSI-CELP) coding method, it notes performing LPC composition. Both linear predictor coefficients alpha and the source of excitation are broadband-ized, and there are broadband-ized alpha and a thing which performs LPC composition by the source of excitation.
[0005] However, distortion is included in the wideband voice obtained by this. Then, in the frequency component contained in the Hara voice, since it is [with the Hara voice] naturally more nearly quality, the filter removed this component among the compounded wideband voices, and how to add the Hara voice has been taken.

[Translation done.]

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EFFECT OF THE INVENTION

[Effect of the Invention] According to this invention, the wideband voice suitable for liking of a user can be offered by making adjustable the frequency characteristics of a high-frequency component, for example, gain, and a frequency band.

[Translation done.]

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TECHNICAL PROBLEM

[Problem(s) to be Solved by the Invention] By the way, although it was the wideband voice compounded as mentioned above, the favorite individual difference of tone quality was large, and the gain of the high-frequency component by which guess composition was carried out was understood that how to bend to a fixed value is good. Similarly, a high-frequency component 6kHz or more has desirable how depending on which this value also bends to immobilization, although the way oppressed a little is liked.
[0007] This invention is made in view of the above-mentioned actual condition, and aims at offer of the bandwidth escape approach and equipment which can adjust the frequency characteristics of a high-frequency component according to liking of a user.

[Translation done.]

MEANS

[Means for Solving the Problem] Since how to add the Hara voice and the compounded component out of band about gain is taken, it becomes possible by adjusting the gain of a component out of band before addition. Moreover, it becomes possible by covering the filter which adjusts frequency characteristics before addition or after addition about bandwidth.
[0009] For this reason, in the bandwidth escape approach which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, after the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band and which can be changed adjusts the bandwidth escape approach of this invention, it is added to the above-mentioned narrow-band signal.
[0010] Moreover, in the bandwidth escape approach which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, the parameter value which was able to be given beforehand and which can be changed adjusts the frequency characteristics of the above-mentioned component out of band after being added to the above-mentioned narrow-band signal.
[0011] Furthermore, the bandwidth growth equipment of this invention is equipped with the frequency-characteristics adjustment device adjusted with the parameter value which was able to give beforehand the frequency characteristics of the above-mentioned component out of band, and which can be changed, and an addition means add the component out of band to which frequency characteristics were adjusted with the above-mentioned frequency-characteristics adjustment device to the above-mentioned narrow-band signal, in the bandwidth growth equipment which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this.
[0012] moreover, the above of the addition outputs of an addition means add the above-mentioned component out of band to the above-mentioned narrow-band signal in the bandwidth growth equipment which guesses a component out of band, adds to the above-mentioned narrow-band signal, and extends bandwidth from the parameter which can compound a narrow-band signal or this, and the above-mentioned addition means -- even if few, it has the frequency-characteristics adjustment device which adjusts with the parameter value which was able to give beforehand the frequency characteristics of one component out of band, and which can be changed.
[0013]
[Embodiment of the Invention] Hereafter, it explains, referring to a drawing about the gestalt of operation of this invention. The gestalt of this operation is speech bandwidth growth equipment which extends the bandwidth of the inputted narrow-band voice, using the bandwidth escape approach concerning this invention. The bandwidth escape approach which this speech bandwidth growth equipment uses is the bandwidth escape approach which guesses a component out of band, adds to the narrow-band signal compounded from the parameter, and extends bandwidth from the parameter which can compound the narrow-band signal restricted in a transmission line, and after the parameter value to which the frequency characteristics of the above-mentioned component out of band were beforehand given by the request of a user and which can be changed adjusts it, it is the approach of adding to the above-mentioned narrow-band signal. For details, it mentions later.
[0014] This speech bandwidth growth equipment is applied to a digital cell phone unit. First, the configuration of this digital cell phone unit is explained. Here, although the transmitter and receiver side is described separately, it is collectively built in one cell phone unit in fact.
[0015] In a transmitter side, the sound signal inputted from the microphone 1 is changed into a digital signal with A/D converter 2, after encoding with the voice encoder 3, transmitting processing is performed to an output bit with a transmitter 4, and it transmits from an antenna 5.
[0016] At this time, the voice encoder 3 supplies the coding parameter in consideration of narrow-band-ization restricted by the transmission line to a transmitter 4. For example, as a coding parameter, there are a parameter about the source of excitation and linear predictor coefficients alpha.
[0017] Moreover, in a receiver side, a receiver 7 receives the electric wave caught with the antenna 6. And the above-mentioned coding parameter is decoded with the voice decryption vessel 8, and voice is extended using the above-mentioned decryption parameter with speech bandwidth growth equipment 9. Then, it returns to an analog sound signal with D/A converter 10, and outputs from a loudspeaker 11.
[0018] The 1st example of the above-mentioned speech bandwidth growth equipment 9 in this digital cell phone unit is shown in drawing 2 . The speech bandwidth growth equipment 9 shown in this drawing 2 extends audio bandwidth using the coding parameter sent from the voice encoder 3 of the transmitting side of the above-mentioned digital cell phone unit.

[0019] The above-mentioned coding parameter is decoded with the voice decryption vessel 8. If the coding approach in a voice coder 3 is based on a PSI-CELP (Pitch Synchronus Innovation-CELP: source of pitch synchronous noise excitation-CELP) coding method, the decryption approach in this voice decryption machine 8 is also depended on PSI-CELP.
[0020] The parameter about the source of excitation which is the 1st coding parameter of the above-mentioned coding parameters decoded with the voice decryption vessel 8 is supplied to the zero stuffing section 12. Moreover, the linear predictor coefficients alpha which are the 2nd coding parameter of the above-mentioned coding parameters are supplied to the alpha->r (linear predictor coefficients -> autocorrelation) conversion circuit 13. Moreover, the signal decoded with the voice decryption vessel 8 is supplied to the V/UV judging circuit 14.
[0021] Moreover, speech bandwidth growth equipment 9 is equipped with the code book 15 for broadband voiced sounds and the code book 16 for broadband silent sounds which are beforehand created using the object for voiced sounds and the parameter for non-vocal sound which were extracted from a broadband voiced sound besides the zero stuffing section 12, the alpha->r conversion circuit 13, and the V/UV judging circuit 14, and non-vocal sound.
[0022] Furthermore, the partial extract circuit 17 and the partial extract circuit 18 which this speech bandwidth growth equipment 9 carries out the partial extract of each code vector in the code book 15 for broadband voiced sounds, and the code book 16 for broadband silent sounds, and ask for a narrow-band parameter, The quantizer 19 for narrow-band voiced sounds which quantizes the autocorrelation for narrow-band voiced sounds from the alpha->r conversion circuit 13 using the narrow-band parameter from the partial extract circuit 17, The quantizer 20 for narrow-band silent sounds which quantizes the autocorrelation for narrow-band silent sounds from the above-mentioned alpha->r conversion circuit 13 using the narrow-band parameter from the partial extract circuit 18, The reverse quantizer 21 for broadband owner vocal sound which reverse-quantizes the narrow-band voiced sound dosage child-ized data from the quantizer 19 for narrow-band voiced sounds using the code book 15 for broadband voiced sounds, The reverse quantizer 22 for broadband non-vocal sound which reverse-quantizes the narrow-band silent sound dosage child-ized data from the quantizer 20 for narrow-band silent sounds using the code book 16 for broadband silent sounds, While changing the autocorrelation for broadband voiced sounds used as the reverse quantization data from the reverse quantizer 21 for broadband owner vocal sound into the linear predictor coefficients for broadband voiced sounds The autocorrelation which changes the autocorrelation for broadband silent sounds used as the reverse quantization data from the reverse quantizer 22 for broadband non-vocal sound into the linear predictor coefficients for broadband silent sounds -> The linear-predictor-coefficients (r->alpha) conversion circuit 23, It comes to have the LPC composition circuit 24 which compounds wideband voice based on the linear predictor coefficients for broadband voiced sounds from this r->alpha conversion circuit 23, the linear predictor coefficients for broadband silent sounds, and the source of excitation from the zero stuffing section 12.
[0023] Moreover, this speech bandwidth growth equipment 9 is equipped with the band stop filter (BSF) 25 which removes the signal component of 300Hz - 3400Hz of frequency bands of input narrow-band voice data from the synthetic output from the rise sample circuit 25 which carries out over sampling technique of the sampling frequency of the narrow-band voice data decrypted with the voice decryption vessel 8 to 16kHz from 8kHz, and the LPC composition circuit 24.
[0024] Furthermore, this speech bandwidth growth equipment 9 is equipped with the frequency-characteristics controller 26 which adjusts the frequency characteristics of the high frequency component 3400Hz or more from BSF25 with the parameter value which was able to be given beforehand, and which can be changed, and the adder 31 which adds the frequency component 3400Hz or more to which frequency characteristics were adjusted by this frequency-characteristics controller 26 to the narrow-band voice data component of the origin of 300Hz - 3400Hz of frequency bands from the above-mentioned rise sample circuit 25.
[0025] And from an output terminal 32, a frequency band is 300-7000Hz, and the digital sound signal whose sampling frequency is 16kHz is outputted.
[0026] Here, the frequency-characteristics controller 26 adjusts the frequency band of the above-mentioned component out of band with the high region oppression filter 27. The high region oppression filter 27 shall be a filter which oppresses component about 6kHz or more, and shall tend to hear the above-mentioned component out of band. The filter factor maintenance memory 28 is connected to the high region oppression filter 27. Some filter factors which make attenuation of frequency characteristics gently-sloping, or are made steep are memorized by this filter factor maintenance memory 28. These filter factors are chosen according to actuation by the user on a control unit 33. And with the high region oppression filter 27, the frequency band of a component out of band is adjusted using the filter factor chosen according to liking of a user.
[0027] Moreover, the frequency-characteristics controller 26 adjusts the gain of the above-mentioned component out of band. The gain set point of the shoes set up beforehand is specifically memorized in the gain set point memory 30, it chooses according to the request of a user in a control unit 33, and a multiplier 29 is supplied. For this reason, in a multiplier 29, the gain of the above-mentioned component out of band can be adjusted according to a request of a user.
[0028] On the whole, this speech bandwidth growth equipment 9 operates as follows. First, a broadband parameter is presumed from a narrow-band parameter, and the wideband voice signal is searched for in the LPC composition circuit 24. And the low-pass side which is a frequency band of the Hara voice is permuted by the Hara voice after that. That is, using BSF25 as a high pass filter, it leaves only a high region, and also in this high-frequency component, a high frequency component is oppressed with the high region oppression filter 27, gain is further adjusted in the signal-processing section 29, and it is adding to the Hara voice.
[0029] Two, broadband-izing of alpha and broadband-izing of the source of excitation, are required for presumption of a broadband parameter. Moreover, it is necessary to create beforehand the code book by the autocorrelation r which is a parameter convertible into alpha and mutual for broadband-ization of alpha. Autocorrelation r is broadband-ized by

quantization by this code book, and reverse quantization.
[0030] First, broadband-ization of alpha is explained. alpha is once changed into the autocorrelation r which is a parameter showing another spectral envelope which is easy to presume a high region side paying attention to being a filter factor showing a spectral envelope, broadband-izes this, and it carries out inverse transformation to alphaw from the broadband autocorrelation rw after that. Vector quantization is used for an escape. What is necessary is to vector-quantize the narrow-band autocorrelation rn and just to calculate rw which corresponds from the index.
[0031] Since fixed relation is realized so that it may mention later, that what is necessary is to prepare only the code book by the broadband autocorrelation, in a narrow-band autocorrelation and a broadband autocorrelation, a narrow-band autocorrelation can be vector-quantized by this, and a broadband autocorrelation can be found by reverse quantization in them.
[0032] About a narrow-band signal, there is relation which shows a broadband signal in the following (1) types at the band-limited thing, then a broadband autocorrelation and a narrow-band autocorrelation.
[0033]
[Equation 1]

$$\phi(x_n)=\phi(x_w\otimes h)=\phi(x_w)\otimes\phi(h) \qquad \cdots (1)$$

[0034] Here, for phi, an autocorrelation and xn are [a broadband signal and h of a narrow-band signal and xw] the impulse responses of a band limit filter.
[0035] Furthermore, the following (2) types are obtained from the relation between an autocorrelation and a power spectrum.

[0036]
[Equation 2]

$$\phi(h)=F^{-1}\left(|H|^2\right) \qquad \cdots (2)$$

[0037] Considering another band limit filter with frequency characteristics equal to the power characteristics of this band limit filter, H', then the above-mentioned (2) formula become like the following (3) types about this.
[0038]
[Equation 3]

$$\phi(h)=F^{-1}\left(|H|^2\right)=F^{-1}(H')=h' \qquad \cdots (3)$$

[0039] The pass band of this new filter and the inhibition zone are equivalent to the original band limit filter, and a damping property serves as a square. Therefore, this new filter can also be called band limit filter. Consideration of this simplifies a narrow-band autocorrelation with what band-limited, the convolution, i.e., the broadband autocorrelation, of a broadband autocorrelation and the impulse response of the filter of a band limit. That is, it becomes the following (4) types.
[0040]
[Equation 4]

$$\phi(x_n)=\phi(x_w)\otimes h' \qquad \cdots (4)$$

[0041] As mentioned above, if only a broadband code book is prepared in vector-quantizing a narrow-band autocorrelation, a narrow-band vector required at the time of quantization can be created by the operation, and does not need to prepare a code book beforehand from a narrow-band autocorrelation.
[0042] Furthermore, since each rw code vector has monotone reduction or the curve fluctuated gently-sloping, even if it makes it low-pass by H', there is no big change, and a direct rw code book can perform rn quantization. However, since a sampling frequency is 1/2, it is necessary to compare every other order.
[0043] By dividing into a voiced sound (V) and non-vocal sound (UV), since the still more accurate escape is possible, this is also performing the escape of alpha. In connection with this, the code book also uses two for the object for V, and UV.
[0044] Next, the escape of the source of excitation is explained. In PSI-CELP, the rise sample of the source of excitation in a narrow-band is carried out by inserting a zero value in the zero stuffing section 12, and what generated aliasing distortion is used. Although this approach is very simple, since the power of the original voice and the difference of harmonic structure are saved, it can be said that it is quality sufficient as a source of excitation.
[0045] And Broadband alpha and the source of broadband excitation which were obtained above perform LPC composition in the LPC composition circuit 24.
[0046] Moreover, since the voice by which broadband LPC composition was carried out is of inferior quality as [this], a low-pass side is permuted with the original voice SNDN of a codec output. For this reason, 3.4kHz or more of composite tone is extracted, the rise sample of the codec output is carried out to fs=16kHz by one side, and these are added.
[0047] At this time, adjustment of the gain which carries out multiplication to a high region side is enabled according to liking of a user with the multiplier 29 of the frequency-characteristics controller 26. Since the individual difference for every user is large, this value is made adjustable. That is, the value of high region side gain is beforehand set up by the input from a user, and multiplication is performed with reference to this value.

[0048] Moreover, there is the sound which is easy to hear by giving filtering which oppresses component about 6kHz or more a little with the high region oppression filter 27 of the frequency-characteristics controller 26 to a high region side before addition. This filter factor is selectable according to liking of a user. By processing with the high region oppression filter 27 using the selected filter factor, the frequency band by the side of a high region was made selectable according to liking.
[0049] However, since the processing using this high region oppression filter 26 does not affect the power characteristics by the side of low-pass, it may be performed to the component out of band under addition output of an adder 31. That is, the high region oppression filter 27 of the frequency-characteristics controller 26 may be formed in the latter part of an adder 31. Or it is also possible to give, after adding the filter which dares have influence also on a low-pass side. Wideband voice is obtained by the above.
[0050] Next, detailed actuation of this speech bandwidth growth equipment 9 is explained using the flow chart of drawing 3 .
[0051] The alpha->r conversion circuit 13 changes into Autocorrelation r the linear predictor coefficients alpha decoded with the voice decryption vessel 8 at step S1. Moreover, the signal decoded with the voice decryption vessel 8 is decoded by the V/UV judging circuit 14 at step S2, and distinction of V/UV is performed.
[0052] If a voiced sound / non-vocal sound judging flag is judged at this step S2 to be V, the switch SW which changes the output from the alpha->r conversion circuit 13 will be connected to the narrow-band voiced phonon-ized circuit 19. Moreover, if judged with UV, Switch SW will connect the output from the alpha->r conversion circuit 13 to the narrow-band silent phonon-ized circuit 20.
[0053] When UV judging circuit 14 judges the above-mentioned voiced sound / non-vocal sound judging flag to be V, in step S4, the autocorrelation r for voiced sounds from Switch SW is supplied to the narrow-band V quantization circuit 19, and it quantizes. This quantization uses the parameter for narrow-band V for which it asked at step S3 by the partial extract circuit 17 as mentioned above.
[0054] On the other hand, at step S3, when UV judging circuit 14 judges the above-mentioned voiced sound / non-vocal sound judging flag to be UV, although the autocorrelation r for non-vocal sound from Switch SW is supplied to the narrow-band UV quantization circuit 20 and it quantizes, it quantizes also here in the partial extract circuit 18 using the parameter for narrow-band UV for which it asked by the operation.
[0055] And it reverse-quantizes using the broadband V code book 15 or the broadband UV code book 16 by the broadband V reverse quantization circuit 21 or the broadband UV reverse quantization circuit 22 which corresponds, respectively at step S5, and, thereby, a broadband autocorrelation is obtained.
[0056] And a broadband autocorrelation is changed into alpha by the r->alpha conversion circuit 23 at step S6.
[0057] On the other hand, the rise sample of the parameter about the source of excitation from the voice decryption machine 8 is carried out by zero being packed by the zero stuffing section 12 between samples at step S7, and it is broadband-ized by aliasing. And this is supplied to the LPC composition circuit 24 as a source of broadband excitation.
[0058] And at step S8, the LPC composition circuit 24 carries out LPC composition of Broadband alpha and the source of broadband excitation, and the sound signal of a broadband is acquired.
[0059] However, since it does not pass to the broadband signal searched for by prediction the way things stand but the error by prediction is included, it is of inferior quality. It is better to use the original voice SNDN of a codec output (input voice) as it is about the frequency range of input narrow-band voice especially.
[0060] Therefore, filtering by step S9 of 300-3400Hz of frequency ranges of input narrow-band voice using BSF25 removes among the composite tone from the LPC composition circuit 24.
[0061] And it adds with an adder 29 at step S13 with what carried out the rise sample of the above-mentioned original voice SNDN by the rise sample circuit 25 at step S10. At this time, there is the sound which is easy to hear by filtering with the high region oppression filter 27 which oppresses component about 6kHz or more a little to a high region side at step S11. This filter factor is made selectable as mentioned above.
[0062] Furthermore, at step S12, adjustment of high region side gain is enabled according to liking of a user using the multiplier 29.
[0063] In addition, the creation of a code book used with speech bandwidth growth equipment 9 is explained here.
[0064] Creation of a code book is an approach by GLA (Generalized Lloyd Algorithm) generally known well. Fixed time amount for every 20msec(s), for example, a frame, is asked for wideband voice, and it asks for the autocorrelation to 1 Sadaji, for example, the 6th order, for every break and its frame. The autocorrelation for every frame of this is made into training data, and a 6-dimensional code book is created. At this time, distinction of a voiced sound and non-vocal sound may be performed, the autocorrelation of a voiced sound and the autocorrelations of non-vocal sound may be collected separately, and each code book may be created. In this case, although a code book is referred to during band-spreading processing at the time of the escape of alpha, also at this time, distinction of a voiced sound and non-vocal sound is performed, and a corresponding code book is used.
[0065] With speech bandwidth growth equipment 9, although the code book 12 for broadband voiced sounds and the code book 14 for broadband silent sounds are used, the creation is explained to a detail, referring to drawing 4 and drawing 5 .
[0066] First, a wideband voice signal is prepared for study and framing is carried out to one-frame 20msec(s) at step S31. Next, the classification of a voiced sound (V) and non-vocal sound (UV) is performed by investigating frame energy, the value of a zero cross, etc. in each frame at step S32.
[0067] And in a broadband voiced sound frame, the autocorrelation parameter r to the 6th order is calculated at step S33. Moreover, at step S34, it asks for the autocorrelation parameter r to the 6th order in a broadband silent sound frame.
[0068] From the 6th autocorrelation parameter of each of this frame, a broadband parameter is extracted at step S41 of

drawing 5 , and the broadband V (UV) code book of a dimension 6 is created at step S42 by GLA.

[0069] As mentioned above, with the speech bandwidth growth equipment using the decryption approach by PSI-CELP, the wideband voice which a user likes mutually can be offered by making adjustable high region gain and a high region oppression filter.

[0070] Next, it explains, referring to drawing 6 about the 2nd example of the above-mentioned speech bandwidth growth equipment. Since it is equipment with which this 2nd example also extends speech bandwidth using the coding parameter sent from the voice encoder 3 of the transmitting side of the above-mentioned digital cell phone unit, the decryption according to the coding approach in the voice encoder 3 is performed.

[0071] If the coding approach in a voice coder 3 is based on a VSELP (Vector Sum Excited Linear Prediction: vector sum exciting line form prediction) coding method, the decryption approach in the voice decryption machine 8 of the preceding paragraph of this speech bandwidth growth equipment is also depended on VSELP.

[0072] The parameter about the source of excitation which is the 1st coding parameter of the above-mentioned coding parameters decoded with the voice decryption vessel 8 is supplied to the source switch section 36 of excitation of drawing 6 . Moreover, the linear predictor coefficients alpha which are the 2nd coding parameter of the above-mentioned coding parameters are supplied to the alpha->r (linear predictor coefficients -> autocorrelation) conversion circuit 13. Moreover, the decoded signal is supplied to the V/UV judging circuit 14.

[0073] Differing from the speech bandwidth growth equipment using PSI-CELP shown in above-mentioned drawing 2 is the point of having established the source switch circuit 36 of excitation in the preceding paragraph of the zero stuffing section 12.

[0074] Although PSI-CELP is performing the codec itself and processing which can be smoothly heard on audibility especially in V, when there is this [no] in VSELP, for this reason a bandwidth escape is carried out, as the noise mixed a little, it can be heard. Then, in case the source of broadband excitation is created, processing like drawing 7 is performed by the source switch circuit 36 of excitation. Processing here is only changed with the processing which showed step S87 - step S89 to above-mentioned drawing 3 .

[0075] The source of excitation of VSELP is beta by the parameter beta (long-term prediction coefficient), bL [i] (long-term filter condition), and gamma (gain) used for a codec, and c1 [i] (excitation code vector). * bL [i] + gamma Although created as * c1[i] Among these, since the former expresses a pitch component and the latter expresses a noise component, it is beta about this. * bL [i] and gamma It divides into * c1[i]. At step S87 In the fixed time amount range, since a pitch was considered to be a strong voiced sound when the former energy is large, it progressed to YES at step S88, and the source of excitation was made into the pulse train, and in the part without a pitch component, it progressed to NO and oppressed to 0. moreover, the step S87 -- case energy is not large -- as usual -- ** -- it considered as the source of broadband excitation by carrying out, and the zero stuffing section's 12 stuffing 0 like PSI-CELP, and carrying out a rise sample to the source of narrow-band excitation created in this way at step S89. Thereby, the quality on the audibility of the voiced sound in VSELP improved.

[0076] If this processing is written by software, it will become like the following (5) types.

[0077]

[Equation 5]

$$
\begin{array}{l}
if\left(\sum_i(\beta * bL[i])^2 > \sum_i(\gamma * c1[i])^2\right)\{ \\
\quad if\left(\beta * bL[i] > \left|Max(\beta * bL[i])\right|\right)\{ \\
\qquad exc_{wide}[2i] = \beta * bL[i]; \\
\quad \}else\{ \\
\qquad exc_{wide}[2i] = 0; \\
\quad \} \\
\}else\{ \\
\quad exc_{wide}[2i] = \beta * bL[i] + \gamma * c1[i]; \\
\}
\end{array}
\quad \cdots (5)
$$

[0078] And it adds with an adder 31 at step S13 with what carried out the rise sample of the above-mentioned original voice SNDN by the rise sample circuit 25 at step S92. At this time, there is the sound which is easy to hear by filtering with the high region oppression filter 27 which oppresses component about 6kHz or more a little to a high region side at step S94. This filter factor supposes that it is selectable, as mentioned above.

[0079] Furthermore, at step S95, adjustment of high region side gain is enabled according to liking of a user using the multiplier 29.

[0080] In addition, this invention is not limited only to what predicts a high region from low-pass. In a means to predict a broadband spectrum, a signal is not restricted to voice.

[0081] Moreover, also when reproducing the signal accumulated in package media with a regenerative apparatus and extending bandwidth, it can apply.

[Translation done.]

* NOTICES *

DESCRIPTION OF DRAWINGS

[Brief Description of the Drawings]
[Drawing 1] It is the block diagram of the digital cell phone unit with which the speech bandwidth growth equipment used as the gestalt of operation of this invention is applied.
[Drawing 2] It is the block diagram of the 1st example of the above-mentioned speech bandwidth growth equipment.
[Drawing 3] It is a flow chart for explaining actuation of the 1st example of the above-mentioned speech bandwidth growth equipment.
[Drawing 4] It is a flow chart for explaining the training-data generation processing used for the code book used by the 1st example of the above-mentioned speech bandwidth growth equipment.
[Drawing 5] It is a flow chart for explaining generation of the above-mentioned code book.
[Drawing 6] It is the block diagram of the 2nd example of the above-mentioned speech bandwidth growth equipment.
[Drawing 7] It is a flow chart for explaining actuation of the 2nd example of the above-mentioned speech bandwidth growth equipment.
[Description of Notations]
8 Voice Decryption Machine, 9 Speech Bandwidth Growth Equipment, 12 Zero Stuffing Section, 13 Linear predictor coefficients -> an autocorrelation (alpha->r) conversion circuit, 14 Voiced sound V / non-vocal sound UV judging circuit, 15 The code book for broadband voiced sounds, 16 The code book for broadband silent sounds, 17 A partial extract circuit, 18 A partial extract circuit, 19 The quantizer for narrow-band voiced sounds, The quantizer for 20 narrow-band silent sounds, 21 The reverse quantizer for broadband owner vocal sound, 22 The reverse quantizer for broadband non-vocal sound, 23 Autocorrelation -> [Linear-predictor-coefficients (r->alpha) conversion circuit,] 24LPC composition circuit, 25 A band stop filter (BSF), 26 A frequency-characteristics controller, 27 A quantity region oppression filter, 28 Filter factor memory, 29 A multiplier, 30 Gain set point memory

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[Translation done.]

* NOTICES *

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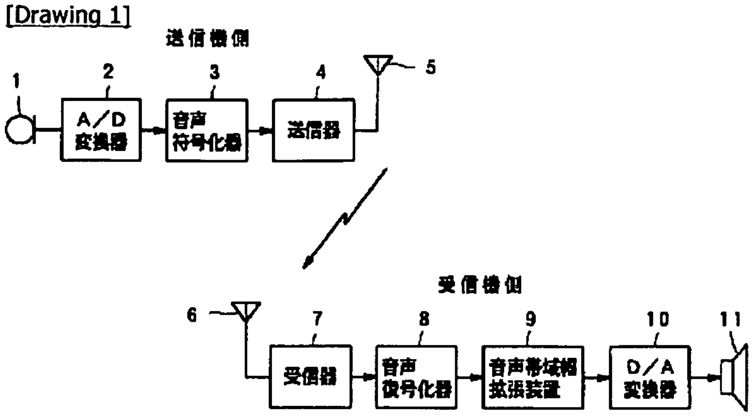
damages caused by the use of this translation.

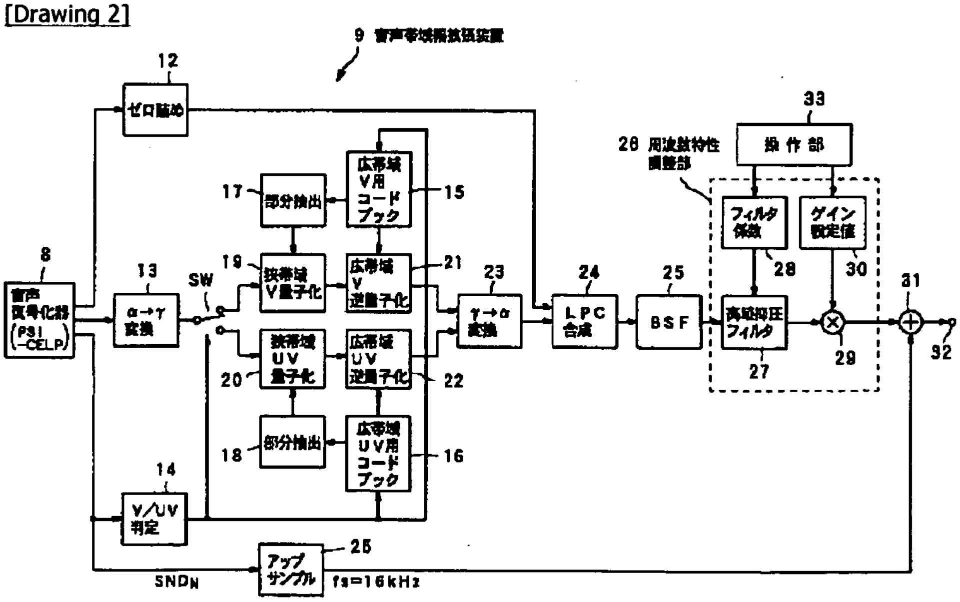
1.This document has been translated by computer. So the translation may not reflect the original precisely.
2.**** shows the word which can not be translated.
3.In the drawings, any words are not translated.

DRAWINGS

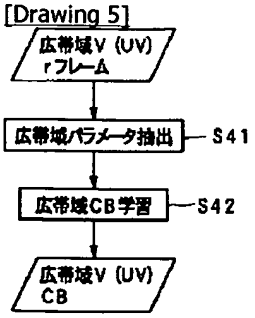
[Drawing 1]



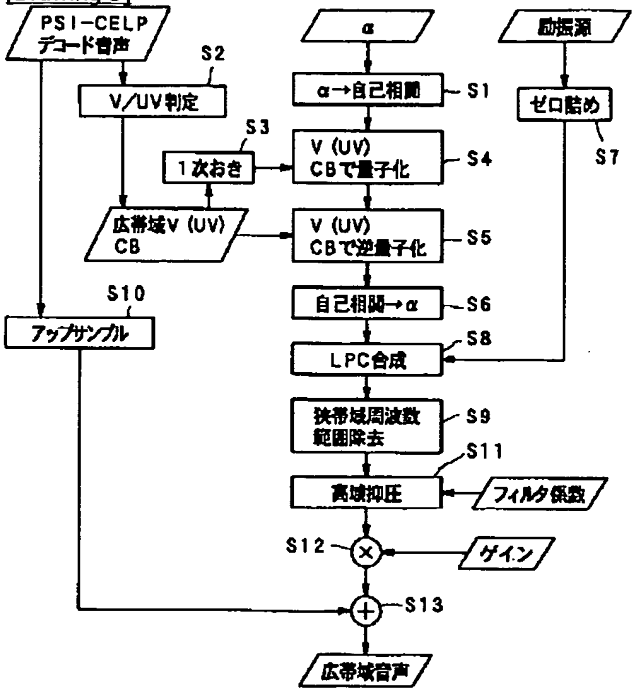
[Drawing 2]



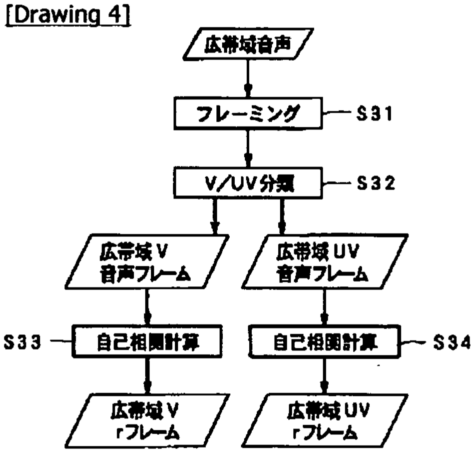
[Drawing 5]



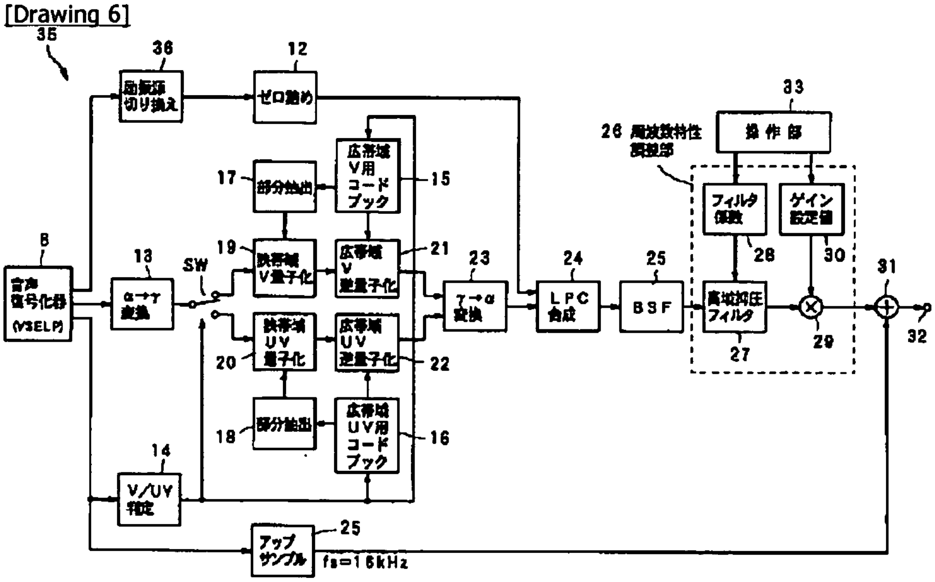
[Drawing 3]



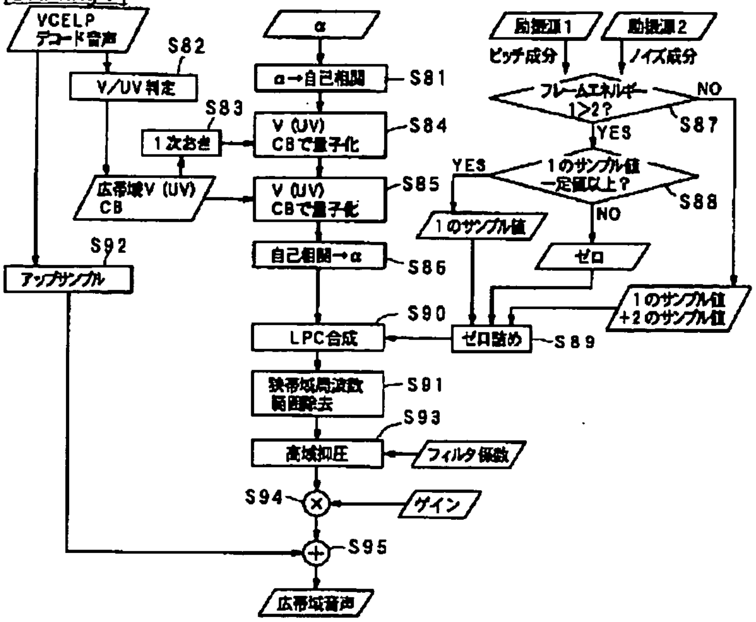
[Drawing 4]



[Drawing 6]



[Drawing 7]



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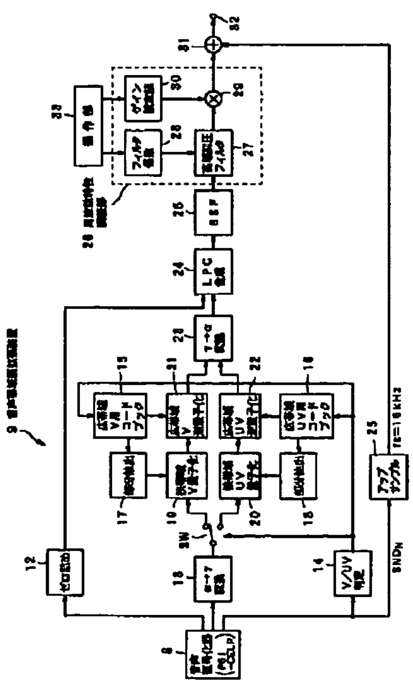
最終頁に続く

(54)【発明の名称】　帯域幅拡張方法及び装置

(57)【要約】

【課題】　広帯域信号の高域成分の周波数特性をユーザの好みに合わせて調整することができる帯域幅拡張方法及び装置を提供する。

【解決手段】　音声帯域幅拡張装置9において、周波数特性調整部26は、BSF25からの3400Hz以上の高い周波数成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整する。加算器31は、周波数特性調整部26で周波数特性が調整された3400Hz以上の周波数成分をアップサンプル回路25からの周波数帯域300Hz～3400Hzの元の狭帯域音声成分に加算する。



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【特許請求の範囲】

【請求項１】　狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張方法において、

上記帯域外成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整してから上記狭帯域信号に加算することを特徴とする帯域幅拡張方法。

【請求項２】　上記周波数特性の調整として上記帯域外成分のゲインを調整することを特徴とする請求項１記載の帯域幅拡張方法。

【請求項３】　上記周波数特性の調整として上記帯域外成分の周波数帯域を調整することを特徴とする請求項１記載の帯域幅拡張方法。

【請求項４】　狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張方法において、

上記狭帯域信号に加算された後の上記帯域外成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整することを特徴とする帯域幅拡張方法。

【請求項５】　上記周波数特性の調整として上記帯域外成分の周波数帯域を調整することを特徴とする請求項４記載の帯域幅拡張方法。

【請求項６】　狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張装置において、

上記帯域外成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整する周波数特性調整手段と、

上記周波数特性調整手段で周波数特性が調整された帯域外成分を上記狭帯域信号に加算する加算手段とを備えることを特徴とする帯域幅拡張装置。

【請求項７】　上記周波数特性調整手段は、上記帯域外成分のゲインを調整することを特徴とする請求項６記載の帯域幅拡張装置。

【請求項８】　上記周波数特性調整手段は、予め与えられた変更可能なゲインン設定値を上記帯域外成分に乗算することを特徴とする請求項７記載の帯域幅拡張装置。

【請求項９】　上記周波数特性調整手段は、上記帯域外成分の周波数帯域を調整することを特徴とする請求項６記載の帯域幅拡張装置。

【請求項１０】　上記周波数特性調整手段は、予め与えられら変更可能なフィルタ係数に基づいて上記帯域外成分の周波数帯域を調整することを特徴とする請求項９記載の帯域幅拡張装置。

【請求項１１】　狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張装置において、

上記帯域外成分を上記狭帯域信号に加算する加算手段と、

上記加算手段の加算出力の内の、上記帯域外成分の周波数特性を予め与えられた変更可能なパラメータ値によって調整する周波数特性調整手段とを備えることを特徴とする帯域幅拡張装置。

【請求項１２】　上記周波数特性調整手段は、上記加算手段の加算出力の内の、上記帯域外成分の周波数帯域を調整することを特徴とする請求項１１記載の帯域幅拡張装置。

【請求項１３】　上記周波数特性調整手段は、上記加算手段の加算出力の内の、上記帯域外成分の周波数帯域を、予め与えられた変更可能なフィルタ係数に基づいて調整することを特徴とする請求項１２記載の帯域幅拡張装置。

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【発明の詳細な説明】

【０００１】

【発明の属する技術分野】本発明は、通信、放送によって伝えられる周波数帯域の狭い音声信号またはそれを構成するパラメータを、送信、伝送路ではそのままにし、受信側で帯域幅を拡張して広帯域音声信号にする帯域幅拡張方法及び装置に関する。また、パッケージメディアに蓄積された信号の帯域幅を拡張して広帯域信号とする帯域幅拡張方法及び装置に関する。

【０００２】

【従来の技術】電話回線の帯域は例えば３００〜３４００Ｈｚと狭く、電話回線を介して送られてくる音声信号の周波数帯域は制限されている。このため、従来のアナログ電話回線の音質はあまり良好とは言えない。また、ディジタル携帯電話の音質についても不満がある。

【０００３】しかしながら、伝送路の規格が定まっているため、この帯域幅を広げることは難しい。したがって、受信側で帯域外の信号成分を予測し、広帯域信号を生成するシステムが様々提案されている。

【０００４】中でも、我が国の自動車／携帯電話の音声コーデック方式であるベクトル和励起線形予測（Vector Sum Excited Linear Prediction：ＶＳＥＬＰ）符号化、ピッチ同期雑音励振源−符号励起線形予測（Pitch Synchronus Innovation−CodeExited Linear Prediction：ＰＳＩ−ＣＥＬＰ）符号化方式に適用を試みた方式では、ＬＰＣ合成を行うことに着目し、線形予測係数αと励振源の両方を広帯域化し、広帯域化されたαと励振源によりＬＰＣ合成を行うものがある。

【０００５】しかしながら、これによって得られた広帯域音声には歪みが含まれる。そこで、原音声に含まれていた周波数成分においては、当然原音声のままの方が品質は良いので、合成された広帯域音声のうちこの成分をフィルタにより除去し、原音声を加算するという手法を取っている。

【０００６】

【発明が解決しようとする課題】ところで、以上のように合成された広帯域音声であるが、音質の好みの個人差が大きく、推測合成された高域成分のゲインは固定値にしない方が良いことが分かった。同様に、６ＫＨｚ以上の高域成分は若干抑圧したほうが好まれるが、この値も固定にしない方が好ましい。

【０００７】本発明は、上記実情に鑑みてなされたものであり、高域成分の周波数特性をユーザの好みに合わせて調整することのできる帯域幅拡張方法及び装置の提供を目的とする。

【０００８】

【課題を解決するための手段】ゲインに関しては、原音声と合成された帯域外成分を加算するという手法を取っているために、加算前に帯域外成分のゲインを調整することで可能となる。また帯域幅に関しては、加算前もしくは加算後に周波数特性を調整するフィルタをかけることで可能となる。

【０００９】このため、本発明の帯域幅拡張方法は、狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張方法において、上記帯域外成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整してから上記狭帯域信号に加算する。

【００１０】また、狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張方法において、上記狭帯域信号に加算された後の上記帯域外成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整する。

【００１１】さらに、本発明の帯域幅拡張装置は、狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張装置において、上記帯域外成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整する周波数特性調整手段と、上記周波数特性調整手段で周波数特性が調整された帯域外成分を上記狭帯域信号に加算する加算手段とを備える。

【００１２】また、狭帯域信号もしくはこれを合成することが可能なパラメータから、帯域外成分を推測し、上記狭帯域信号に加算して帯域幅を拡張する帯域幅拡張装置において、上記帯域外成分を上記狭帯域信号に加算する加算手段と、上記加算手段の加算出力の内の、上記少なくとも一つの帯域外成分の周波数特性を予め与えられた変更可能なパラメータ値によって調整する周波数特性調整手段とを備える。

【００１３】

【発明の実施の形態】以下、本発明の実施の形態について図面を参照しながら説明する。この実施の形態は、本発明に係る帯域幅拡張方法を用いながら、入力された狭帯域音声の帯域幅を拡張する音声帯域幅拡張装置である。この音声帯域幅拡張装置が用いる帯域幅拡張方法は、伝送路で制限される狭帯域信号を合成することが可能なパラメータから、帯域外成分を推測し、パラメータから合成した狭帯域信号に加算して帯域幅を拡張する帯域幅拡張方法であり、上記帯域外成分の周波数特性を、ユーザの所望により予め与えられた変更可能なパラメータ値によって調整してから上記狭帯域信号に加算するという方法である。詳細については後述する。

【００１４】この音声帯域幅拡張装置は、ディジタル携帯電話装置に適用される。先ず、このディジタル携帯電話装置の構成を説明しておく。ここでは、送信機側と受信機側を別々に記しているが、実際には一つの携帯電話装置内にまとめて内蔵されている。

【００１５】送信機側では、マイクロホン１から入力された音声信号を、Ａ／Ｄ変換器２によりディジタル信号に変換し、音声符号化器３により符号化してから送信器４で出力ビットに送信処理を施し、アンテナ５から送信する。

【００１６】このとき、音声符号化器３は、伝送路により制限される狭帯域化を考慮した符号化パラメータを送信器４に供給する。例えば、符号化パラメータとしては、励振源に関するパラメータや、線形予測係数αがある。

【００１７】また、受信機側では、アンテナ６で捉えた電波を、受信器７で受信する。そして、音声復号化器８で上記符号化パラメータを復号し、音声帯域幅拡張装置９で上記復号化パラメータを用いて音声を拡張する。その後、Ｄ／Ａ変換器１０でアナログ音声信号に戻して、スピーカ１１から出力する。

【００１８】このディジタル携帯電話装置における、上記音声帯域幅拡張装置９の第１の具体例を図２に示す。この図２に示す音声帯域幅拡張装置９は、上記ディジタル携帯電話装置の送信側の音声符号化器３から送られてきた符号化パラメータを用いて音声の帯域幅を拡張する。

【００１９】上記符号化パラメータは音声復号化器８により復号される。音声符号器３での符号化方法がＰＳＩ－ＣＥＬＰ（Pitch Synchronus Innovation - CELP：ピッチ同期雑音励振源－ＣＥＬＰ）符号化方式によるものであるとすれば、この音声復号化器８での復号化方法もＰＳＩ－ＣＥＬＰによる。

【００２０】音声復号化器８で復号された、上記符号化パラメータの内の第１の符号化パラメータである励振源に関するパラメータは、ゼロ詰め部１２に供給される。また、上記符号化パラメータの内の第２の符号化パラメータである線形予測係数αはα→ｒ（線形予測係数→自己相関）変換回路１３に供給される。また、音声復号化器８で復号された信号は、Ｖ／ＵＶ判定回路１４に供給

される。

【0021】また、音声帯域幅拡張装置9は、ゼロ詰め部12と、α→r変換回路13と、V／UV判定回路14の他、広帯域有声音及び無声音から抽出した有声音用及び無声音用パラメータを用いて予め作成されている広帯域有声音用コードブック15と広帯域無声音用コードブック16とを備える。

【0022】さらに、この音声帯域幅拡張装置9は、広帯域有声音用コードブック15と広帯域無声音用コードブック16内の各コードベクトルを部分抽出して狭帯域パラメータを求める部分抽出回路17及び部分抽出回路18と、α→r変換回路13からの狭帯域有声音用自己相関を部分抽出回路17からの狭帯域パラメータを用いて量子化する狭帯域有声音用量子化器19と、上記α→r変換回路13からの狭帯域無声音用自己相関を部分抽出回路18からの狭帯域パラメータを用いて量子化する狭帯域無声音用量子化器20と、狭帯域有声音用量子化器19からの狭帯域有声音用量子化データを広帯域有声音用コードブック15を用いて逆量子化する広帯域有声音用逆量子化器21と、狭帯域無声音用量子化器20からの狭帯域無声音用量子化データを広帯域無声音用コードブック16を用いて逆量子化する広帯域無声音用逆量子化器22と、広帯域有声音用逆量子化器21からの逆量子化データとなる広帯域有声音用自己相関を広帯域有声音用の線形予測係数に変換すると共に広帯域無声音用逆量子化器22からの逆量子化データとなる広帯域無声音用自己相関を広帯域無声音用の線形予測係数に変換する自己相関→線形予測係数（r→α）変換回路23と、このr→α変換回路23からの広帯域有声音用線形予測係数と広帯域無声音用線形予測係数とゼロ詰め部12からの励振源とに基づいて広帯域音声を合成するLPC合成回路24とを備えてなる。

【0023】また、この音声帯域幅拡張装置9は、音声復号化器8で復号化された狭帯域音声データのサンプリング周波数を8kHzから16kHzにオーバーサンプリングするアップサンプル回路25と、LPC合成回路24からの合成出力から入力狭帯域音声データの周波数帯域300Hz～3400Hzの信号成分を除去するバンドストップフィルタ（BSF）25を備えている。

【0024】さらに、この音声帯域幅拡張装置9は、BSF25からの3400Hz以上の高い周波数成分の周波数特性を、予め与えられた変更可能なパラメータ値によって調整する周波数特性調整部26と、この周波数特性調整部26で周波数特性が調整された3400Hz以上の周波数成分を上記アップサンプル回路25からの周波数帯域300Hz～3400Hzの元の狭帯域音声データ成分に加算する加算器31とを備えている。

【0025】そして、出力端子32からは、周波数帯域が300～7000Hzで、サンプリング周波数が16kHzのディジタル音声信号が出力される。

【0026】ここで、周波数特性調整部26は、上記帯域外成分の周波数帯域を高域抑圧フィルタ27で調整する。高域抑圧フィルタ27は、例えば約6KHz以上の成分を抑圧するフィルタで、上記帯域外成分を聴きやすいものとする。高域抑圧フィルタ27にはフィルタ係数保持メモリ28が接続されている。このフィルタ係数保持メモリ28には、周波数特性の減衰をなだらかにしたり、急峻にしたりするフィルタ係数がいくつか記憶されている。これらのフィルタ係数は、操作部33上でのユーザによる操作に応じて選択される。そして、高域抑圧フィルタ27では、ユーザの好みに応じて選択されたフィルタ係数を用いて帯域外成分の周波数帯域を調整する。

【0027】また、周波数特性調整部26は、上記帯域外成分のゲインを調整する。具体的には、予め設定されたいくつかのゲイン設定値をゲイン設定値メモリ30に記憶しておき、操作部33におけるユーザの所望に応じて選択して乗算器29に供給する。このため、乗算器29では、ユーザの所望に応じて、上記帯域外成分のゲインを調整することができる。

【0028】この音声帯域幅拡張装置9は、全体的に以下のように動作する。先ず、狭帯域パラメータから広帯域パラメータを推定し、LPC合成回路24で広帯域音声信号を求めている。そして、その後、原音声の周波数帯域である低域側を原音声に置換する。すなわち、高域通過フィルタとしてBSF25を用い、高域のみを残し、この高域成分の中でも高い周波数成分を高域抑圧フィルタ27で抑圧し、さらに信号処理部29でゲインを調整し、原音声に加算している。

【0029】広帯域パラメータの推定は、αの広帯域化、励振源の広帯域化の二つが必要である。また、αの広帯域化には、αと相互に変換可能なパラメータである自己相関rによるコードブックを予め作成しておく必要がある。このコードブックによる量子化、逆量子化によって自己相関rが広帯域化される。

【0030】先ず、αの広帯域化について説明する。αはスペクトル包絡を表すフィルタ係数であることに着目し、高域側を推定しやすい別のスペクトル包絡を表すパラメータである自己相関rに一旦変換し、これを広帯域化し、その後で広帯域自己相関rwからαwに逆変換する。拡張にはベクトル量子化を用いる。狭帯域自己相関rnをベクトル量子化し、そのインデックスから対応するrwを求めればよい。

【0031】狭帯域自己相関と広帯域自己相関には、後述するように一定の関係が成り立つため、広帯域自己相関によるコードブックのみを用意すればよく、狭帯域自己相関をこれによりベクトル量子化でき、また逆量子化により広帯域自己相関が求まる。

【0032】狭帯域信号を、広帯域信号を帯域制限したものとすれば、広帯域自己相関と狭帯域自己相関には以

下の（１）式に示す関係がある。

【００３３】

＊【数１】

$$\phi(x_n)=\phi(x_w\otimes h)=\phi(x_w)\otimes\phi(h) \quad \cdots(1)$$

【００３４】ここで、φは自己相関、xnは狭帯域信号、xwは広帯域信号、ｈは帯域制限フィルタのインパルス応答である。

【００３５】さらに、自己相関とパワースペクトルの関係から、次の（２）式が得られる。

【００３６】

【数２】

$$\phi(h)=F^{-1}\left(|H|^2\right) \quad \cdots(2)$$

※【００３７】この帯域制限フィルタのパワー特性と等しい周波数特性を持つ、もう一つの帯域制限フィルタを考え、これをＨ’とすれば、上記（２）式は、次の（３）式のようになる。

【００３８】

【数３】

$$\phi(h)=F^{-1}\left(|H|^2\right)=F^{-1}(H')=h' \quad \cdots(3)$$

【００３９】この新たなフィルタの通過域、阻止域は当初の帯域制限フィルタと同等であり、減衰特性が２乗となる。したがって、この新たなフィルタもまた、帯域制限フィルタといえる。これを考慮すると、狭帯域自己相関は、広帯域自己相関と帯域制限のフィルタのインパル★ス応答との畳み込み、すなわち広帯域自己相関を帯域制限したものと単純化される。すなわち、次の（４）式となる。

【００４０】

【数４】

$$\phi(x_n)=\phi(x_w)\otimes h' \quad \cdots(4)$$

【００４１】以上より、狭帯域自己相関をベクトル量子化するにあたっては、広帯域コードブックのみを用意すれば、量子化時に必要な狭帯域ベクトルは演算により作成が可能であり、狭帯域自己相関から予めコードブックを用意しておく必要がない。

【００４２】さらに、各ｒwコードベクタは単調減少もしくはなだらかに増減するカーブを持つために、Ｈ’により低域通過させても大きな変化がなく、ｒn量子化は、直接ｒwコードブックで行える。ただし、サンプリング周波数が１／２のため、１次おきに比較する必要がある。

【００４３】αの拡張は有声音（Ｖ）と無声音（ＵＶ）に分けることによって、さらに精度良い拡張が可能であるため、これも行っている。これに伴いコードブックもＶ用、ＵＶ用の二つを用いている。

【００４４】次に、励振源の拡張について説明する。ＰＳＩ－ＣＥＬＰにおいては狭帯域での励振源を、ゼロ詰め部１２でゼロ値を挿入することでアップサンプルし、エイリアシング歪みを発生させたものを用いる。この方法は非常に単純であるが、元の音声のパワーや調波構造の差分が保存されるので、励振源としては十分な品質であるといえる。

【００４５】そして、以上で得られた広帯域αと広帯域励振源によりＬＰＣ合成回路２４でＬＰＣ合成を行う。

【００４６】また、広帯域ＬＰＣ合成された音声は、このままでは品質が悪いので、低域側はコーデック出力のオリジナル音声ＳＮＤNで置換する。このために、合成☆音のうち３．４ＫＨｚ以上を抽出し、一方でコーデック出力をｆｓ＝１６ＫＨｚにアップサンプルし、これらを加算する。

【００４７】このとき、周波数特性調整部２６の乗算器２９で高域側に乗算するゲインをユーザの好みに応じて調整可能としている。ユーザ毎の個人差が大きいため、この値を可変にしている。つまり、高域側ゲインの値をユーザからの入力により予め設定しておき、この値を参照し、乗算を行う。

【００４８】また、加算前に高域側に対し、周波数特性調整部２６の高域抑圧フィルタ２７で約６ＫＨｚ以上の成分を若干抑圧するフィルタリングを施すことで、聴きやすい音にしている。このフィルタ係数は、ユーザの好みに応じて選択可能である。選択されたフィルタ係数を用いて高域抑圧フィルタ２７で処理を行うことで、好みに応じ高域側の周波数帯域を選択可能とした。

【００４９】ただし、この高域抑圧フィルタ２６を用いての処理は、低域側のパワー特性に影響を与えないため、加算器３１の加算出力中の帯域外成分に施してもよい。すなわち、加算器３１の後段に、周波数特性調整部２６の高域抑圧フィルタ２７を設けてもよい。あるいは、あえて低域側にも影響のあるフィルタを加算後に施す事も可能である。以上により広帯域音声が得られる。

【００５０】次に、この音声帯域幅拡張装置９の詳細な動作について図３のフローチャートを用いて説明する。

【００５１】ステップＳ１でα→ｒ変換回路１３は、音声復号化器８でデコードされた線形予測係数αを自己相

関ｒに変換する。また、音声復号化器８でデコードされた信号はステップＳ２でＶ／ＵＶ判定回路１４により解読され、Ｖ／ＵＶの判別が行われる。

【００５２】このステップＳ２で有声音／無声音判定フラグがＶと判定されると、$\alpha\rightarrow r$変換回路１３からの出力を切り替えるスイッチＳＷは、狭帯域有声音量子化回路１９に接続する。また、ＵＶと判定されるとスイッチＳＷは、$\alpha\rightarrow r$変換回路１３からの出力を狭帯域無声音量子化回路２０に接続する。

【００５３】ＵＶ判定回路１４が上記有声音／無声音判定フラグをＶと判定したとき、ステップＳ４ではスイッチＳＷからの有声音用自己相関ｒを狭帯域Ｖ量子化回路１９に供給して量子化する。この量子化は、上述したように部分抽出回路１７によりステップＳ３で求めた狭帯域Ｖ用パラメータを用いる。

【００５４】一方、ＵＶ判定回路１４が上記有声音／無声音判定フラグをＵＶと判定したときには、ステップＳ３では、スイッチＳＷからの無声音用自己相関ｒを狭帯域ＵＶ量子化回路２０に供給して量子化するが、ここでも、部分抽出回路１８で演算により求めた狭帯域ＵＶ用パラメータを用いて量子化する。

【００５５】そして、ステップＳ５でそれぞれ対応する広帯域Ｖ逆量子化回路２１又は広帯域ＵＶ逆量子化回路２２により広帯域Ｖコードブック１５又は広帯域ＵＶコードブック１６を用いて逆量子化し、これにより広帯域自己相関が得られる。

【００５６】そして、広帯域自己相関はステップＳ６で$r\rightarrow\alpha$変換回路２３によりαに変換される。

【００５７】一方で、音声復号化器８からの励振源に関するパラメータは、ステップＳ７でゼロ詰め部１２によりサンプル間にゼロが詰められることでアップサンプルされ、エイリアシングにより広帯域化される。そして、これが広帯域励振源として、ＬＰＣ合成回路２４に供給される。

【００５８】そして、ステップＳ８で、ＬＰＣ合成回路２４が広帯域αと広帯域励振源とを、ＬＰＣ合成し、広帯域の音声信号が得られる。

【００５９】しかし、このままでは予測によって求められた広帯域信号にすぎず、予測による誤差が含まれているので品質が悪い。特に入力狭帯域音声の周波数範囲に関しては、コーデック出力のオリジナル音声SND_N（入力音声）をそのまま利用したほうが良い。

【００６０】したがって、ＬＰＣ合成回路２４からの合成音のうち、入力狭帯域音声の周波数範囲３００～３４００ＨｚをステップＳ９でＢＳＦ２５を用いたフィルタリングにより除去する。

【００６１】そして、ステップＳ１０でアップサンプル回路２５により上記オリジナル音声SND_Nをアップサンプルしたものと、ステップＳ１３で加算器２９により加算する。このとき、ステップＳ１１で高域側に対し、約６ＫＨｚ以上の成分を若干抑圧する高域抑圧フィルタ２７によりフィルタリングを施すことで、聴きやすい音にしている。このフィルタ係数は上述したように選択可能とされている。

【００６２】さらに、ステップＳ１２では、乗算器２９を用いてユーザの好みに応じて高域側ゲインを調整可能としている。

【００６３】なおここで、音声帯域幅拡張装置９で用いる、コードブックの作成について説明する。

【００６４】コードブックの作成は一般によく知られたＧＬＡ(Generalized Lloyd Algorithm)による方法である。広帯域音声を一定時間、例えば２０msecごとのフレームに区切り、そのフレーム毎に、一定次例えば６次までの自己相関を求めておく。このフレーム毎の自己相関をトレーニングデータとし、６次元のコードブックを作成する。このとき、有声音、無声音の区別を行い、有声音の自己相関、無声音の自己相関を別々に集め、それぞれのコードブックを作成してもよい。この場合、帯域拡張処理中αの拡張時、コードブックを参照するが、このときにも有声音、無声音の判別を行い、対応するコードブックを利用する。

【００６５】音声帯域幅拡張装置９では、広帯域有声音用コードブック１２と広帯域無声音用コードブック１４を用いているが、図４及び図５を参照しながらその作成について詳細に説明する。

【００６６】先ず、広帯域音声信号を学習用に用意し、ステップＳ３１で１フレーム２０msecにフレーミングする。次に、ステップＳ３２で各フレームにおいて、例えばフレームエネルギーやゼロクロスの値等を調べることによって有声音（Ｖ）か無声音（ＵＶ）かの分類を行う。

【００６７】そして、ステップＳ３３で広帯域有声音フレームにおいて、例えば６次までの自己相関パラメータｒを計算する。また、ステップＳ３４では広帯域無声音フレームにおける、例えば６次までの自己相関パラメータｒを求める。

【００６８】この各フレームの６次の自己相関パラメータから、図５のステップＳ４１で広帯域パラメータを抽出し、ＧＬＡにより次元６の広帯域Ｖ（ＵＶ）コードブックをステップＳ４２で作成する。

【００６９】以上、ＰＳＩ－ＣＥＬＰによる復号化方法を用いた音声帯域幅拡張装置では、高域ゲイン、高域抑圧フィルタを可変とすることで、ユーザの好み合う広帯域音声を提供することができる。

【００７０】次に、上記音声帯域幅拡張装置の第２の具体例について図６を参照しながら説明する。この第２の具体例も、上記ディジタル携帯電話装置の送信側の音声符号化器３から送られてきた符号化パラメータを用いて音声帯域幅を拡張する装置であるため、音声符号化器３での符号化方法に従った復号化を行う。

【００７１】音声符号器３での符号化方法がＶＳＥＬＰ（Vector Sum Excited Linear Prediction：ベクトル和励起線形予測）符号化方式によるものであるとすれば、この音声帯域幅拡張装置の前段の音声復号化器８での復号化方法もＶＳＥＬＰによる。

【００７２】音声復号化器８で復号された、上記符号化パラメータの内の第１の符号化パラメータである励振源に関するパラメータは、図６の励振源切り換え部３６に供給される。また、上記符号化パラメータの内の第２の符号化パラメータである線形予測係数αは$\alpha \rightarrow r$（線形予測係数→自己相関）変換回路１３に供給される。また、復号された信号はＶ／ＵＶ判定回路１４に供給される。

【００７３】上記図2に示したＰＳＩ－ＣＥＬＰを用いた音声帯域幅拡張装置と異なるのは、励振源切り換え回路３６をゼロ詰め部１２の前段に設けている点である。

【００７４】ＰＳＩ－ＣＥＬＰは、コーデック自体、特にＶを聴感上滑らかに聞こえるような処理を行っているが、ＶＳＥＬＰにはこれがなく、このために帯域幅拡張したときに若干雑音が混入したように聞こえる。そこで、広帯域励振源を作成する際に、励振源切り換え回路３６により図７のような処理を施す。ここでの処理は、ステップＳ８７～ステップＳ８９を上記図3に示した処理と異ならせるだけである。

【００７５】ＶＳＥＬＰの励振源は、コーデックに利用されるパラメータβ(長期予測係数)，bL[i](長期フィルタ状態)，γ(利得)，c1[i](励起コードベクタ)により、$\beta * bL[i] + \gamma * c1[i]$として作成されるが、このうち前者がピッチ成分、後者がノイズ成分を表すので、これを$\beta * bL[i]$と$\gamma * c1[i]$に分け、ステップＳ８７で、一定の時間範囲において、前者のエネルギーが大きい場合にはピッチが強い有声音と考えられるため、ステップＳ８８でＹＥＳに進み、励振源をパルス列とし、ピッチ成分のない部分ではＮＯに進み０に抑圧した。また、ステップＳ８７でエネルギーが大きくない場合には従来どおりとし、こうして作成された狭帯域励振源にステップＳ８９でゼロ詰め部１２によりPSI-CELP同様０を詰めアップサンプルすることにより広帯域励振源とした。これにより、ＶＳＥＬＰにおける有声音の聴感上の品質が向上した。

【００７６】この処理をソフトウェア的に書くと以下の（５）式のようになる。

【００７７】

【数５】

$$
\begin{array}{l}
if\left(\sum_i(\beta * bL[i])^2 > \sum_i(\gamma * c1[i])^2\right)\{ \\
\quad if\left(\beta * bL[i] > \left|Max(\beta * bL[i])\right|\right)\{ \\
\qquad exc_{wide}[2i] = \beta * bL[i]; \\
\quad \}else\{ \\
\qquad exc_{wide}[2i] = 0; \\
\quad \} \\
\}else\{ \\
\quad exc_{wide}[2i] = \beta * bL[i] + \gamma * c1[i]; \\
\}
\end{array}
\qquad \cdots (5)
$$

【００７８】そして、ステップＳ９２でアップサンプル回路２５により上記オリジナル音声SND_Nをアップサンプルしたものと、ステップＳ１３で加算器３１により加算する。このとき、ステップＳ９４で高域側に対し、約６ＫＨｚ以上の成分を若干抑圧する高域抑圧フィルタ２７によりフィルタリングを施すことで、聴きやすい音にしている。このフィルタ係数は上述したように選択可能としている。

【００７９】さらに、ステップＳ９５では、乗算器２９を用いてユーザの好みに応じて高域側ゲインを調整可能としている。

【００８０】なお、本発明は低域から高域を予測するものだけに限定するものではない。広帯域スペクトルを予測する手段においては信号を音声に限るものではない。

【００８１】また、パッケージメディアに蓄積された信号を再生装置で再生するときに帯域幅を拡張するときにも適用できる。

【００８２】

【発明の効果】本発明によれば、高域成分の周波数特性、例えばゲイン、周波数帯域を可変とすることで、ユーザの好みに合う広帯域音声を提供することができる。

【図面の簡単な説明】

【図１】本発明の実施の形態となる音声帯域幅拡張装置が適用されるディジタル携帯電話装置のブロック図である。

【図２】上記音声帯域幅拡張装置の第１の具体例のブロック図である。

【図３】上記音声帯域幅拡張装置の第１の具体例の動作を説明するためのフローチャートである。

【図４】上記音声帯域幅拡張装置の第１の具体例で用いられるコードブックに使われるトレーニングデータ生成処理を説明するためのフローチャートである。

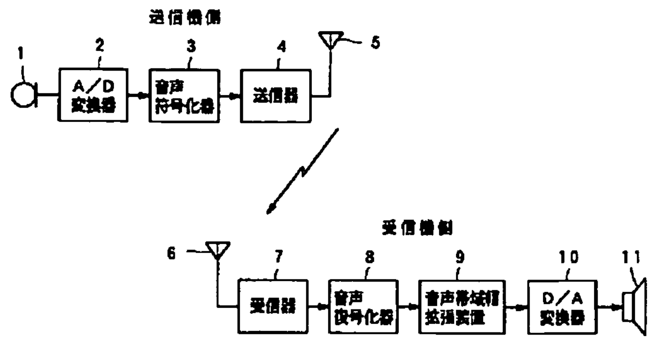
【図５】上記コードブックの生成を説明するためのフローチャートである。

【図６】上記音声帯域幅拡張装置の第２の具体例のブロック図である。

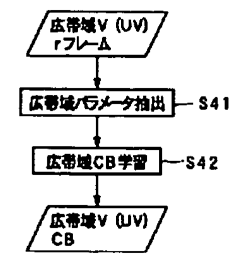
【図７】上記音声帯域幅拡張装置の第２の具体例の動作を説明するためのフローチャートである。

【符号の説明】
8 音声復号化器、9 音声帯域幅拡張装置、12 ゼロ詰め部、13 線形予測係数→自己相関（α→r）変換回路、14 有声音V／無声音UV判定回路、15 広帯域有声音用コードブック、16 広帯域無声音用コードブック、17 部分抽出回路、18 部分抽出回路、19 狭帯域有声音用量子化器、20狭帯域無声音用量子化器、21 広帯域有声音用逆量子化器、22 広帯域無声音用逆量子化器、23 自己相関→線形予測係数（r→α）変換回路、24LPC合成回路、25 バンドストップフィルタ（BSF）、26 周波数特性調整部、27 高域抑圧フィルタ、28 フィルタ係数メモリ、29 乗算器、30 ゲイン設定値メモリ

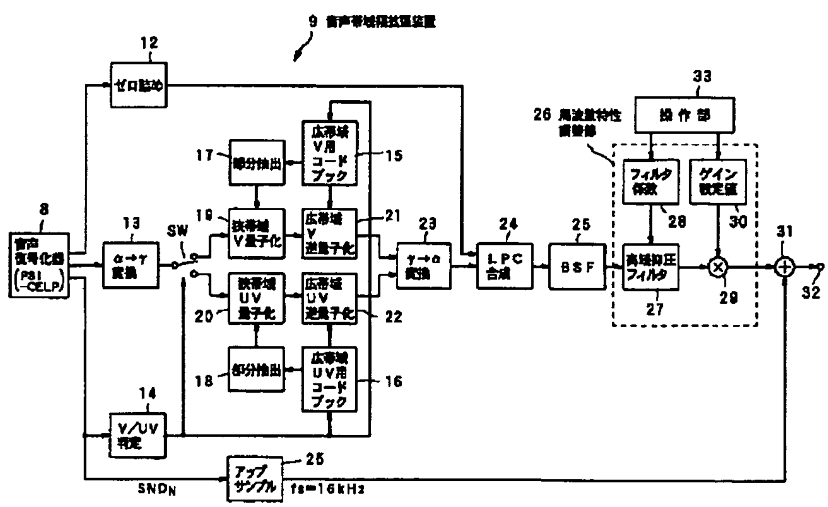
【図1】



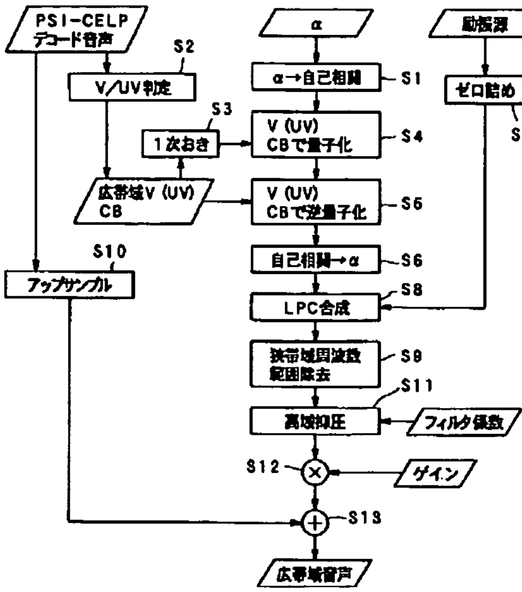
【図5】



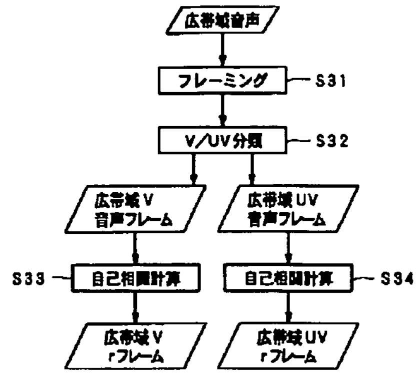
【図2】



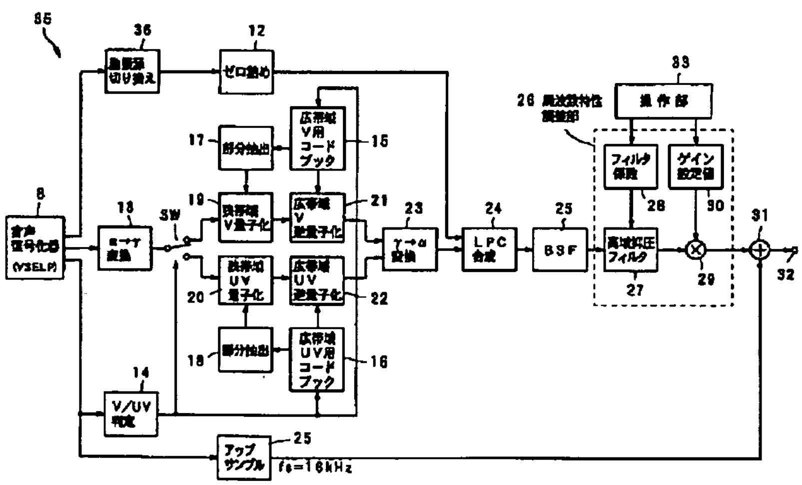
【図3】


PSI−CELP
デコード音声
V/UV判定
S2
1次おき
S3
広帯域V (UV)
CB
S10
アップサンプル
α
α→自己相関
S1
V (UV)
CBで量子化
S4
V (UV)
CBで逆量子化
S5
自己相関→α
S6
LPC合成
S8
狭帯域周波数
範囲除去
S9
S11
高域抑圧
フィルタ係数
S12
ゲイン
S13
広帯域音声
励振源
ゼロ詰め
S7

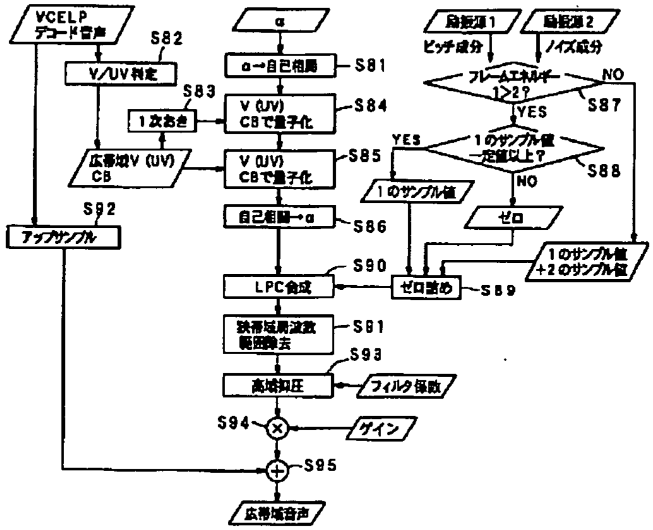
【図4】


広帯域音声
フレーミング
S31
V/UV分類
S32
広帯域V
音声フレーム
広帯域UV
音声フレーム
S33
自己相関計算
自己相関計算
S34
広帯域V
rフレーム
広帯域UV
rフレーム

【図6】


35
36
12
ゼロ詰め
17
部分抽出
広帯域
V用
コード
ブック
15
8
音声
復号化器
(VSELP)
13
α→γ
変換
SW
19
狭帯域
V量子化
広帯域
V
逆量子化
21
23
γ→α
変換
24
LPC
合成
25
BSF
20
狭帯域
UV
量子化
広帯域
UV
逆量子化
22
18
部分抽出
広帯域
UV用
コード
ブック
16
14
V/UV
判定
アップ
サンプル
25
fs=16kHz
33
操作部
26 周波数特性
調整部
フィルタ
係数
28
ゲイン
設定値
30
高域抑圧
フィルタ
27
29
31
32

【図7】



フロントページの続き

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BC08 BC12 BC18 BD02
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HH12 HH22 JJ14
5K066 AA02 BB01 CC02 DD14 DD22
DD32 EE45 JJ03 JJ15